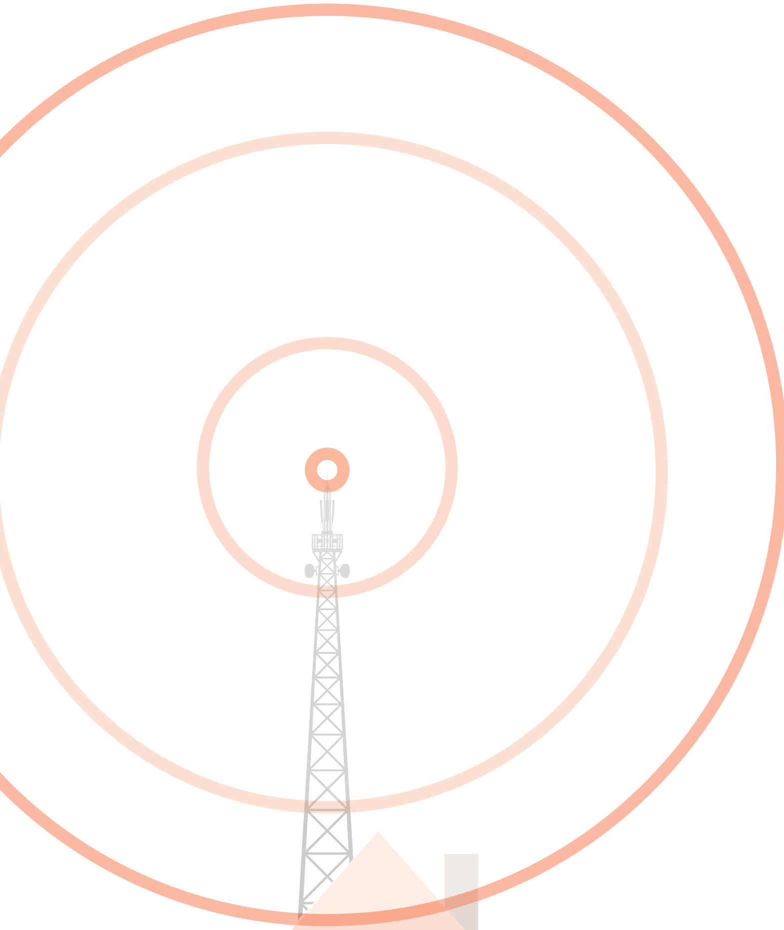


ATSC 3.0

Transition and Implementation GUIDE

Version 1.0

A living guide to assist in the planning of spectrum repack to support ATSC 3.0 implementation

INTRODUCTION

This document was developed to provide broadcasters with ATSC 3.0 information that can inform investment and technical decisions required to move from ATSC 1.0 to ATSC 3.0. It also guides broadcasters who are planning for its adoption while also planning for *channel changes during the FCC Spectrum Repack Program.*

This document, finalized October 11, 2016, will be updated periodically as insight and additional information is made available from industry testing and implementation of the new standard. This document was developed by the companies and organizations listed in the Appendix. Updates to the Guide are open to input from all companies and individuals that wish to contribute. Those interested in suggesting changes or updates to this document can do so at priya.iyer@gatesair.com.

EXECUTIVE SUMMARY

Television service continues to evolve as content distributors – from traditional cable operators to internet-delivered services – utilize the latest technologies to reach viewers and offer a wide variety of program choices.

New receiving devices are easily connected to the internet, which relies on the language of Internet Protocol (IP) to transport content. Now terrestrial broadcasters are preparing both for the adoption of an IP-ready next-generation digital TV (DTV) standard and a realignment of the U.S. TV spectrum.

Viewers are already buying high-quality displays that respond to 4K Ultra HDTV signals and High Dynamic Range (HDR) capabilities. Immersive and personalized audio is also emerging, with the ability to enhance the quality and variety of audio.

To keep pace with these innovations, and to set the stage for additional advances in the future, broadcasters now have the option to move forward with a new broadcast television transmission standard.

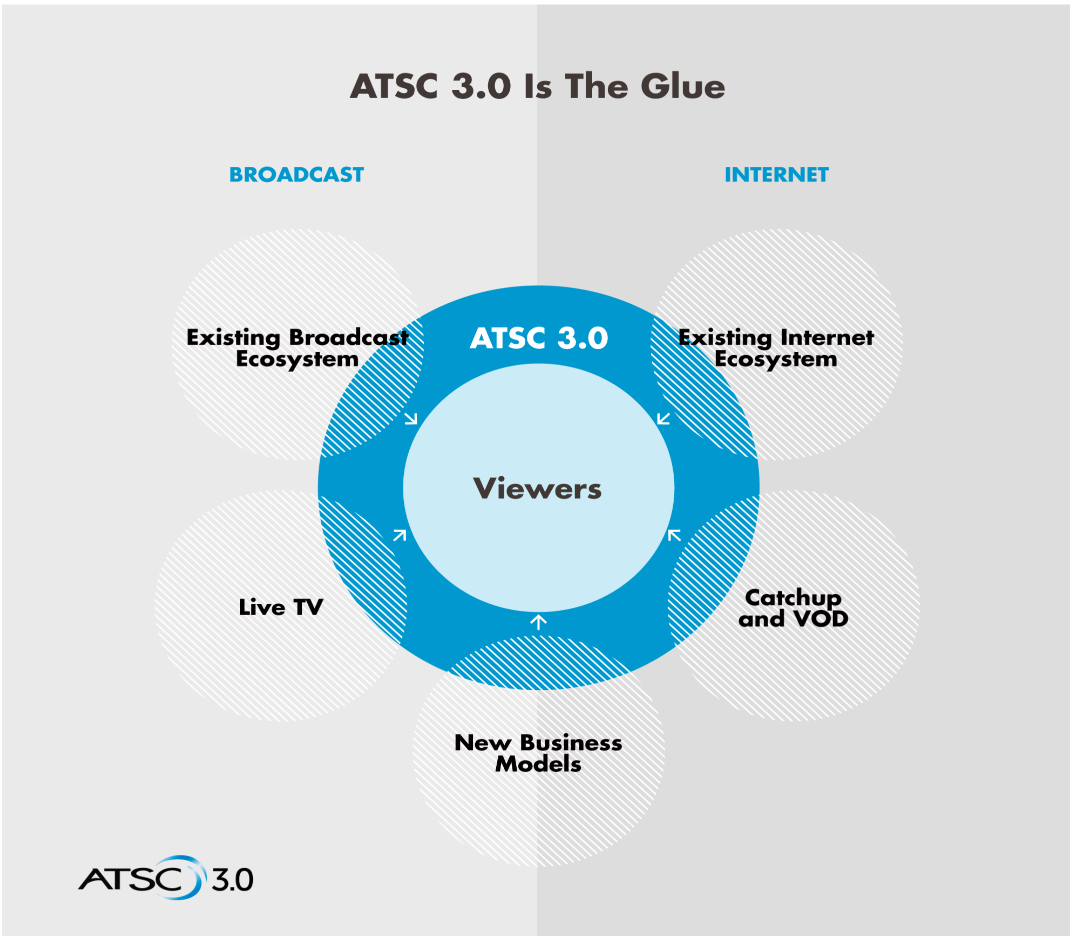
The Advanced Television Systems Committee (ATSC), through a cooperative effort by over 125 member organizations from the broadcast, consumer electronics, cable, satellite, motion picture, professional broadcast equipment, computer, and integrated circuit industries, has developed the ATSC 3.0 television standard – the world's first IP-based broadcast service.

This next-generation TV ATSC 3.0 transmission standard is the "glue" that enables broadcast protocol to exist in an internet environment. The standard will permit broadcasters to offer innovative technologies and services to the public, including:

- Visually stunning pictures on large-screen televisions with superior reception;
- Broadcast programming with multiple consumer-friendly features, such as interactivity and personalized audio, which exceed those available through the current broadcast standard;
- Access to unlimited viewing of local and national news and the most popular sports and entertainment programming, and trusted educational and children's programming, via mobile devices such as tablets and smartphones;
- Seamless integration of broadcast programming with other IP services, with the ability to provide state-of-the-art security that content owners depend upon;
- Advanced emergency alert information backed up with live, professional reporters and connections of public safety officials with the public;

- Datacasting that will offer a new broadband data pipe into the home, thereby giving content providers another means for distributing large video and other digital files to consumers, and providing enhanced opportunities for essential public services including education and public safety; and

- The ability to geotarget news, weather, and other programming.



Next-generation TV transmissions will operate within broadcasters' existing 6 MHz television channel, and be subject to the same radio frequency interference constraints and requirements that apply to the current DTV standard.

No additional spectrum is required, nor has it been requested from the FCC. Next-generation TV services can be deployed within stations' existing coverage contours without causing interference to current digital TV stations.

Because ATSC 3.0 offers significant advancements and capabilities beyond its predecessor, it is not backwards-compatible with the 20-year-old ATSC 1.0 transmission system.

To accomplish a seamless implementation of next-generation TV without disenfranchising viewers, the industry will deploy this new technology in parallel with the existing digital television standard in a voluntary, market-based manner. Parallel implementation will mean that some broadcasters in each market will deploy next-generation ASTC 3.0 TV, while others will continue to transmit using the current ATSC 1.0 standard. With this model, broadcasters in each market will share in order to simulcast their respective signals, so that all viewers can receive programming from their local stations in both the current DTV and new TV formats, all free and over-the-air.

Like mobile carriers today, which are free to choose when and how to deploy new standards, broadcasters will have the option of choosing when and whether to enhance their current service with ATSC 3.0. Broadcasters will also have the option to build Single Frequency Networks (SFNs), which can extend or improve existing coverage, particularly for indoor and mobile device reception. SFNs can also enable geotargeted advertising.

ATSC 3.0 can also unlock new opportunities for broadcasters and their advertising-based business model:

- Targeted Advertising. Channel watermarking allows targeting capabilities, and the IP return path enables the ability to measure audience activity for broadcasters' advertising customers.

- Targeted Content. ATSC 3.0 "targeting" ability also allows broadcasters to better reach narrow audiences with niche content that can increase value for advertisers.

- Subscription Services. New ways to support conditional access for "freemium," one-time, and premium services.

- More Channels. Efficient video and audio compression in the ATSC 3.0 standard will significantly boost effective bandwidth, by a factor of 3 or more, the number of HD program streams a broadcaster can transmit in the current 6 MHz channel of spectrum.

- Mobile Services. Seamless delivery of programs and data services will be specifically targeted to portable and mobile devices.

- Greater Capacity. A system can support a third-party "offloading" business where data, video and other bandwidth-intensive content (such as software updates for devices) can be transmitted over broadcast networks for "edge" storage or delivery to non-household destinations. Today's wireless service and streaming video providers, for example, have acute needs for greater capacity to cost-effectively move their content as close to their customers as possible.

- Second Screen. The ability to deliver program related second-screen content by either Over the Air (OTA), or OTT distribution and interactive content using the IP return channel.

Plan Now for ATSC 3.0: Realize Savings, Maximize Investment and Return

For those broadcasters wanting to take full advantage of ATSC 3.0's many capabilities, early planning is critical. Creating a blueprint of the desired capabilities, their relative importance, and how they will be implemented, is a complex task that will require advance planning from multiple disciplines within a station, or group of stations.

Broadcasters should identify any potential redundancies for tower work or equipment with a future ATSC 3.0 adoption, potentially saving investment costs by not having to do tower work twice – once as the result of the Spectrum Repack and another time to optimize for ATSC 3.0.

This effort can reduce capital requirements in a number of ways:

- Making the right choice of transmitter, RF system and antenna components that will support a future move to ATSC 3.0, even if this means that stations must make an additional investment beyond the FCC repack reimbursement. Purchasing the components that will support stations' future ATSC 3.0 plans can greatly reduce expenses during an ATSC 3.0 transition.

- If a new antenna system must be purchased for a channel change, making sure that the antenna conforms to the RF requirements for ATSC 3.0 adoption.

- If a new transmitter will be required for a channel change, broadcasters should evaluate and pick products that are software-upgradable to ATSC 3.0, and have the ability to easily add additional amplification to support the peak power requirements related to Vpol for ATSC 3.0.

- Eventual installation of ATSC 3.0 equipment could mean changes to tower and tower site infrastructure. If broadcasters incorporate ATSC 3.0 into tower structural engineering studies, tower modifications, and transmitter, RF system and antenna installations during the Spectrum Auction Repack, they will pay only once for potentially expensive and time-consuming work.

What to Consider When Planning for ATSC 3.0 Adoption

Although an ATSC 3.0 system won't necessarily require a wholesale change of equipment and infrastructure, some changes must occur. Keep in mind most new equipment will be more versatile and economically expandable through software updates, thus potentially extending equipment life. Also, the IP-based system can be easily customized and modified for all types of IP-based services, and these modifications can occur so that consumer receivers will not be rendered obsolete.

Many business and technical realities have changed since the ATSC 1.0 digital standard was created. The next-generation platform delivers more flexibility, capabilities, and tradeoffs – depending upon how a system is customized. With that comes a multifaceted system that will require more business and technical planning. Understanding which combination of ATSC 3.0 services to employ and the subsequent trade-offs between robustness and bandwidth is the first step in planning a system.

The following equipment changes will need to be made when implementing the ATSC 3.0 platform:

- ATSC 3.0 specifies HEVC/H.265 video compression, which is two generations ahead of MPEG-2, the ATSC 1.0 codec. New encoders, which include IP encapsulation, will be required.

- Data server systems with software modules supporting service signaling, service management, program and service guide information, and emergency alerting management will be required.

- The new system requires the addition of gateways, where final signals are managed and assembled before they're sent via the Studio to Transmitter Link (STL) to the transmitter.
- Replacement of Studio to Transmitter links with IP-based systems is likely.
- Exciter/modulator replacement with an ATSC 3.0-capable unit is likely, assuming that the current exciter is not software-upgradable.
- Assessment of the usefulness of current transmitters when adopting ATSC 3.0, which employs a different modulation standard (OFDM) than the current system. Implications to consider include maximizing transmitter power output vs. replicating the coverage area and adding Vpol to the antenna to improve mobile reception.
- Because peak power is higher for an ATSC 3.0 system than it is for ATSC 1.0 transmissions, broadcasters will need to carefully assess their antenna systems' needs, assuming that the antenna was not replaced during repack with one that supports the ATSC 3.0 requirements.

Transitioning a TV Market from ATSC 1.0 to ATSC 3.0

The transition from ATSC 1.0 to ATSC 3.0 will be dramatically different than the previous transition from analog to digital TV. Single-market cooperation will be necessary, so that broadcasters can ensure that no viewer loses programming.

One solution suggests a temporary channel-sharing partnership featuring a "Lighthouse Station." One station (the Lighthouse) could seed the market with ATSC 3.0 signals for all stations in a given market, while other stations make unused capacity collectively available to replicate the Lighthouse station's ATSC 1.0 signal, as well as their own ATSC 1.0 signals. Over time, as audiences migrate their viewing to the ATSC 3.0 services, these stations will elect to convert all of their respective transmissions to ATSC 3.0, and no longer transmit an ATSC 1.0 signal.

Eventually, all stations in the market will only transmit ATSC 3.0 signals, but this will only occur after viewers in that market have compatible ATSC 3.0 receivers. Station brand identity will be maintained by logical channel identification that is now present in ATSC 1.0 (as it was with the switch from analog to digital), and will continue to be present in ATSC 3.0.

Consumer receiver availability will be critical, and top TV suppliers such as LG Electronics have already publicly demonstrated ATSC 3.0 prototype receivers in several form factors. Other top TV suppliers, such as Samsung and Sony, have been heavily involved in contributing to the standard. LG, the NAB and others have been involved in developing and demonstrating transition devices, known as home gateways, to enable home reception on smart TVs and IP-enabled devices. In an important development, the Consumer Technology Association (CTA) joined with the NAB and other broadcast industry groups to petition the FCC to authorize use of the next-generation platform.

Conclusion: Start Your Planning Now

The terrestrial TV broadcast community has moved with urgency and focus to finalize the ATSC 3.0 standard in an unprecedented time frame.

The South Korean Ministry of Science, ICT and Future Planning (MSIP) recently selected ATSC 3.0 for its country's Ultra-High-Definition (UHD) television transmission standard that is set to be launched in South Korea in February of 2017. This adoption by South Korea will establish the new standard as an international standard, and comes in time for the carriage of the upcoming XXIII Olympic Winter Games that will take place in PyeongChang, South Korea in February 2018.

The urgency is not solely for the acute need to competitively transform the broadcasting business and related opportunities. Completing the standard so that broadcasters can simultaneously plan for spectrum repacking and ATSC 3.0 implementation can save the industry millions of dollars, and help to “kick start” the transition. With a new spectrum plan for the 600 MHz TV band anticipated in early 2017, planning for ATSC 3.0 adoption now is essential.

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PART 1: A TECHNICAL PRIMER

ATSC 3.0 System Overview

Introduction

ATSC 3.0 is a suite of voluntary technical standards and recommended practices for a digital terrestrial television broadcast system. ATSC 3.0 is fundamentally different from predecessor ATSC systems and is therefore largely incompatible with them. This divergence from earlier design is intended to allow substantial improvements in performance, functionality and efficiency sufficient to warrant implementation of a non-backwards-compatible system. With higher capacity to deliver Ultra High-Definition services, robust reception on a wide range of devices, improved efficiency, interactivity, personalization options and an (IP) -based core, the ATSC 3.0 standard profoundly redefines terrestrial television broadcasting.

Overview

The ATSC 3.0 system is designed with a “layered” architecture due to the many advantages of such a system, particularly pertaining to upgradability and extensibility. This approach follows the Open Systems Interconnection (OSI) model, a conceptual architecture that characterizes and standardizes communications functions found in telecommunications and computer systems. While the canonical OSI model is based on 7 layers, the ATSC 3.0 model combines some of those layers.

A generalized layering model for ATSC 3.0 is shown in Figure below.

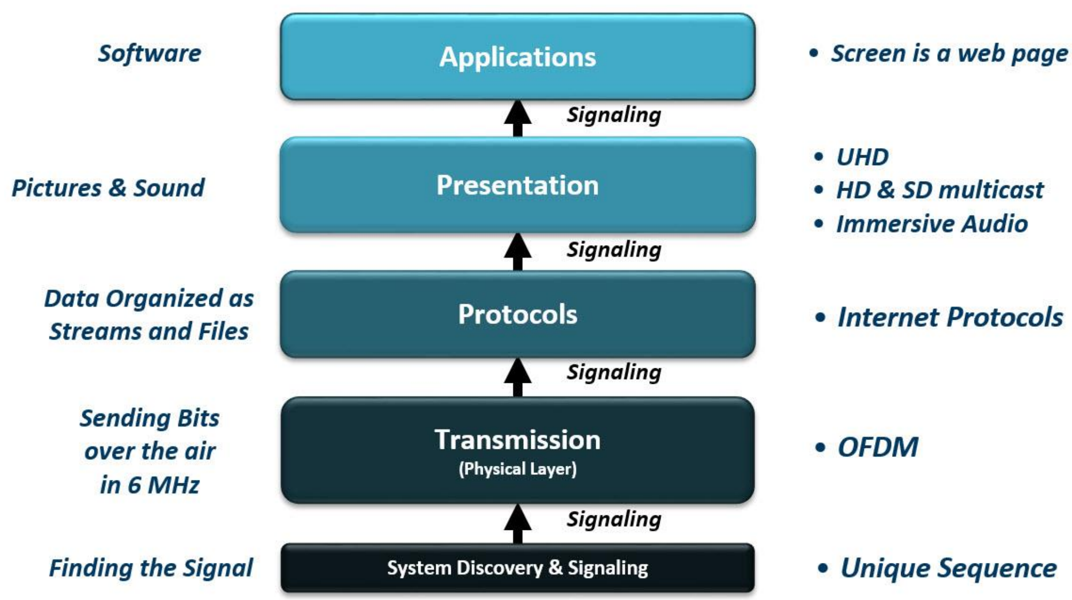

Figure 1: ATSC 3.0 Layered Architecture

The above layering model describes the ATSC 3.0 signal from the perspective of the emitted DTV signal. In contrast, the following ATSC 3.0 overview describes the system from the input of content to the generation of the modulated signal, i.e., the signal flow found in a television broadcast plant, starting with content encoding and ending with the modulated waveform from the exciter.

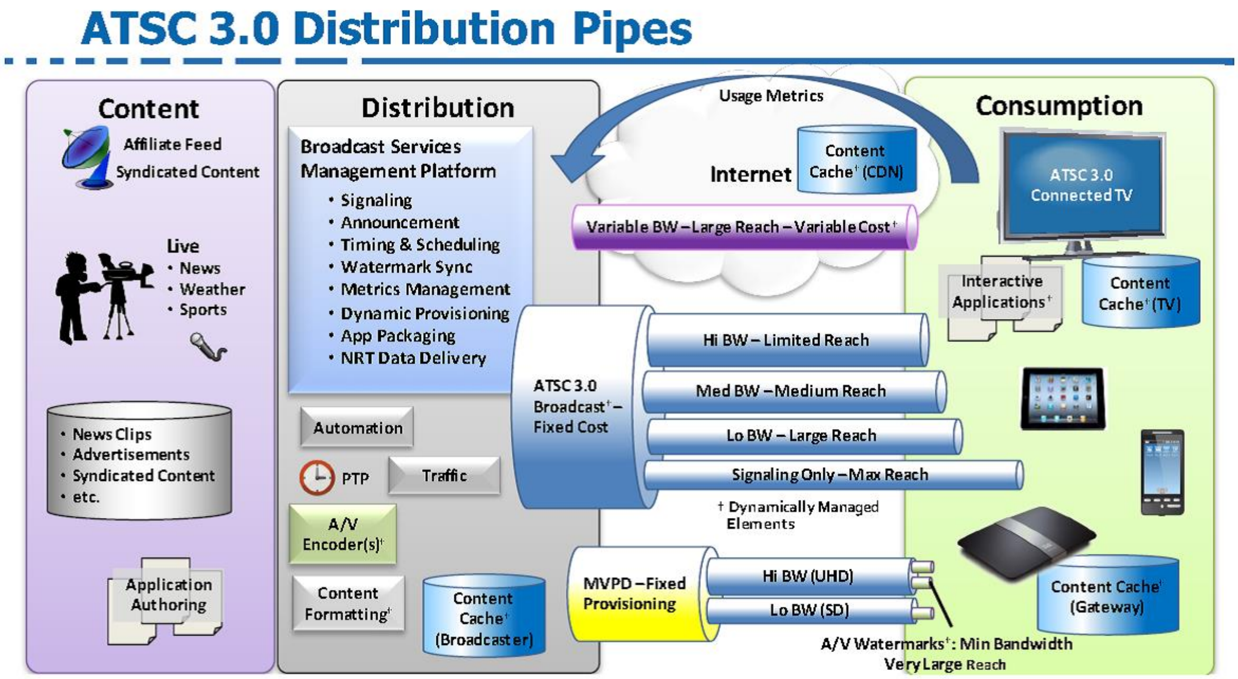

Figure 2: ATSC 3.0 Illustrated System Overview

Source: Triveni Digital

ATSC 3.0 Documentation Structure

The ATSC 3.0 standard is described by a number of separate documents that together comprise the full standard. The documents are presented in this manner to support the independent evolution of different aspects of the standard.

The figure below illustrates the individual documents and the elements of the ATSC 3.0 standard to which they each pertain. It should be noted that some elements of the system span more than one document (e.g., accessibility and emergency alerting).

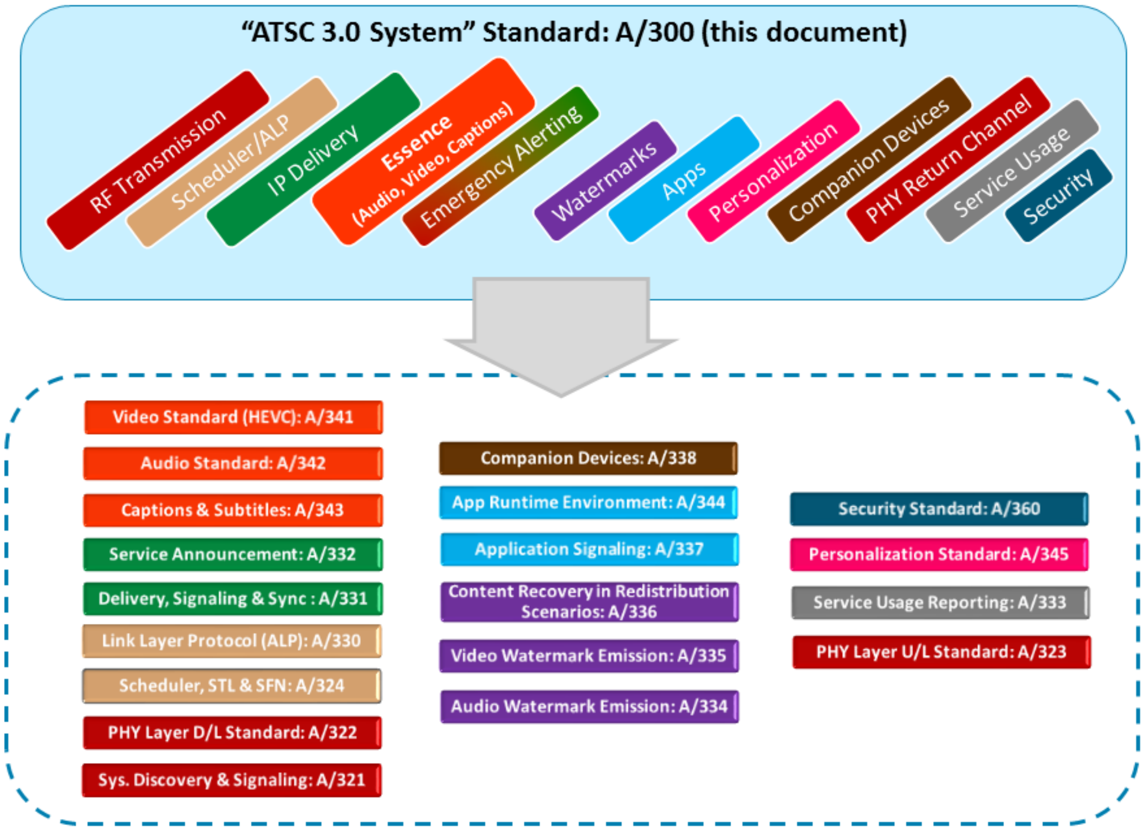

Figure 3: ATSC 3.0 Standards Set and Structure
Source: ATSC Working Draft: ATSC 3.0 System, Doc. S31-204r5, July 18, 2016

Applications and Presentation Layer

The applications and presentation layer focuses on the encoding and presentation of content, which primarily includes video, audio and captioning. (In addition to these three "essence" elements, interactive applications are also part of this layer.)

For this layer, ATSC 3.0 relies on the utilization of the latest, most efficient and feature-rich systems for content presentation. These systems are based on standards that are (or are expected to be) widely utilized on other content delivery platforms.

Video Encoding System

ATSC 3.0 utilizes the MPEG-H Part 2 High Efficiency Video Coding (HEVC), also known as ITU-R H.265. The ATSC 3.0 standard specifies the allowable emission formats of this system, including features such as 4K spatial resolution (up to 2160p), Spatial Scalable Coding (SSC), High Dynamic Range (HDR), Wide Color Gamut (WCG), 3D, and temporal layering.

Legacy formats are included to maximize compatibility with existing content at HD and SD resolutions, including those with interlaced scanning structure and an aspect ratio of 4:3. 3D video is also supported by interlaced and progressive formats.

The ATSC 3.0 HEVC-encoded Legacy SD video formats are encoded with the following constraints:

- The bitstream conforms to HEVC Main 10 Profile, Main Tier, Level 3.1.
- The spatial resolution in both dimensions is evenly divisible by 8.
- In 60 Hz regions, the picture rates of 25 and 50 Hz are not used.
- The color space container is ITU-Recommendation 709.
- The color subsampling is 4:2:0.
- SD resolutions supported include 640, 704 and 720 horizontal pixels in both interlaced and progressive scanning.
- For 720x480 resolution formats, the active 4:3 or 16:9 picture falls within the center 704 pixels. The additional pixels account for the transitions created by analog blanking.

The ATSC 3.0 HEVC-encoded Interlaced HD video formats are encoded with the following constraints:

- The bitstream conforms to HEVC Main 10 Profile, Main Tier, Level 4.1.
- The spatial resolution in both dimensions must be evenly divisible by 8.
- In 60 Hz regions, the picture rates of 25 and 50 Hz are not be used.
- The color space container is ITU Recommendation 709.
- The color subsampling is 4:2:0.
- The HD interlaced formats supported include both 1440x1080 and 1920x1080 pixels.
- These formats are coded with a vertical size of 544 lines per field (1088 lines per frame) in order for the vertical resolution of each picture to be divisible by 8. The bottom 4 lines (8 lines per frame) are black.
- When telecine content is encoded, an inverse telecine process may be applied by the encoder, yielding a coded bitstream of 24 or 24/1.001 Hz progressive 1080x1920 or 1080x1440 pictures.
- The picture rates supported in 60 Hz regions are the following in Hz: 24/1.001, 24, 30/1.001, 30, 60/1.001, 60, 120/1.001, 120.

The ATSC 3.0 HEVC-encoded progressive video formats are supported with the following constraints:

- The spatial resolution is limited to not more than 2160 lines and 3840 horizontal pixels.
- The spatial resolution in both dimensions must be evenly divisible by 8.
- The picture rates supported in 60 Hz regions are the following in Hz: 24/1.001, 24, 30/1.001, 30, 60/1.001, 60, 120/1.001, 120.

- The pixel aspect ratio is 1:1 (square pixels).
- The bitstream will conform to HEVC Main 10 Profile or HEVC Scalable Main 10 Profile, Main Tier, Level 5.2.
- The color space container is ITU Recommendation 709 or ITU Recommendation 2020.
- The color subsampling is 4:2:0.

Coded representation of video with 1080 lines (e.g., 1080x1920) may be coded either as 1080 lines or as 1088 lines. When the video is coded as 1088 lines, the bottom 8 lines are presented as black.

Active Field Descriptor (AFD) and Bar Data are defined such that the active area of the picture does not necessarily need to fill the entire coded area. When the active image area of the emitted video signal does not fill the entire encoded video frame (e.g., when the video is letterboxed or pillarboxed), AFD and Bar Data information should be present in the original source video signal in accordance with SMPTE ST 2016-1 and should be present in the emitted video signal. AFD information and Bar Data are used by receivers to optimize the display of images that do not fill the coded frame.

Audio Encoding System

The ATSC 3.0 audio system provides immersive and personalized sound for television. Because the audio system is a different encoding system when compared with existing 5.1 channel-based systems, it is not compatible with the audio system used in ATSC 1.0 service. The system supports delivery of audio content from mono, stereo, 5.1 channel and 7.1 channel audio sources, as well as from sources supporting immersive audio represented by channel-based audio up to 22.2 channels, object-based audio, or scene-based audio using Higher-Order Ambisonics (HOA). Such representations might not directly map to loudspeaker feeds, but instead could represent the overall sound field.

The ATSC 3.0 audio system enables multiple configurations, which allow flexibility for user personalization, as well as loudness control and various accessibility features:

- Immersive audio is enabled on a wide range of loudspeaker configurations, including loudspeaker configurations with suboptimum loudspeaker locations, and on headphones. The system also enables audio reproduction on loudspeaker configurations not designed for immersive audio such as 7.1 channel, 5.1 channel, two-channel and single-channel loudspeaker systems.
- User control is enabled for certain aspects of the sound scene that is rendered from the encoded representation (e.g., relative level of dialog, music, effects, or other elements important to the user). User-selectable alternative audio tracks can be delivered via terrestrial broadcast or via broadband, and in realtime or non-realtime delivery modes. Such audio tracks may be used to replace the primary audio track, or be mixed with the primary audio track and delivered for synchronous presentation with the corresponding video content.
- Receiver mixing of alternative audio tracks is enabled (e.g., presentation of assistive audio services, other language dialog, special commentary, music and effects, etc.) with the main audio track or other audio tracks, with relative levels and position in the sound field adjustable by the user.

- Broadcasters can provide users with the option of varying the loudness of a TV program's dialog relative to other elements of the audio mix to increase dialog intelligibility.
- Support is provided to normalize and control loudness of reproduced audio content, including adapting the loudness and dynamic range of audio content, as appropriate, for the receiving device and presentation environment.
- Support is provided for inclusion and signaling of audio (speech) that presents an aural representation of emergency information provided by broadcasters via on-screen text display (static, scrolling or "crawling" text).

The ATSC 3.0 audio system establishes a common framework for multiple Next Generation Audio (NGA) systems, both current and future. For North American broadcast stations, the AC-4 audio system is the NGA system specified in ATSC 3.0. ATSC 3.0 also specifies MPEG-H 3D Audio for use in other parts of the world. Constraints on both of these audio standards for their use in ATSC 3.0 are specified within the ATSC 3.0 Audio standard (A/342).

Captions and Subtitles

ATSC 3.0 incorporates the required technology for closed caption and subtitle tracks delivered over Real-Time Object Delivery over Unidirectional Transport (ROUTE)-MPEG, Dynamic Adaptive Streaming over Hyper-Text Transfer Protocol (DASH) and MPEG Media Transport (MMT) transports. This includes the content essence, the packaging and timing, and the transport-dependent signaling.

The caption and subtitle system is based on World Wide Web Consortium (W3C) Internet Media Subtitles and Captions Version 1 (MSC1), an Extensible Markup Language (XML)-based representation of captions. XML is inherently extensible and can be enhanced over time by ATSC retaining compatibility with earlier versions. For example, user systems can extend it using their own namespaces and retain compatibility with the core feature set that is shown below. There are currently no ATSC-defined namespaces or schemas.

The caption and subtitle technology is SMPTE Timed Text (SMPTE-TT) as defined in SMPTE 2052-1. SMPTE-TT was chosen because it:

- Supports world-wide language and symbol tables (specifically including non-Latin);
- Supports world-wide image glyph delivery;
- Is in use today by various "media delivery silos", including broadcaster OTT delivery;
- Is U.S. FCC closed caption safe harbor for IP-delivered content;
- Supports FCC requirements for both 708 and IP captions;
- Is compatible with DECE (UltraViolet) Common File Format Timed Text (CFF-TT).

SMPTE-TT is complex and not all of it is required to meet the closed captions and subtitle requirements of ATSC 3.0. A simpler subset is used for practical implementation. Therefore, W3C's new "Timed Text Markup Language (TTML) Text and Image Profiles for IMSC1" is selected, having been designed specifically for needs like broadcast as well as broadband delivery. In summary:

- Superset of DECE/Ultraviolet CFF-TT (TTML + SMPTE-TT extensions);
- Superset of European Broadcasting Union (EBU)-TT-D being deployed in Europe;
- Two profiles are included:
  - Text Profile requiring a font rendering engine in the decoder;
  - Image Profile with Portable Network Graphics (PNG) files.

The rough feature relationships of the TTML profiles mentioned above are shown in Figure 4.

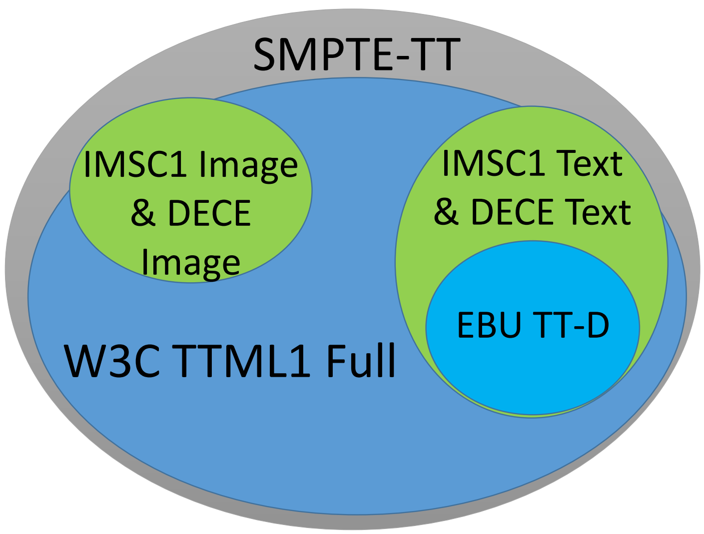

Figure 4: Venn Diagram of TTML profiles

Source: ATSC Candidate Standard: Captions and Subtitles A/343, Doc. S34-169r8, 20 June 2016

For broadband delivery, the DASH segment size should be less than 500K bytes, to reduce the amount of decoder memory needed to decode a document, and also provide a reasonable startup acquisition time at the beginning of a program.

For pre-recorded broadcast, caption ISO/BMFF segments (i.e., IMSC1 documents) should be relatively short in duration to allow decoders to join an in-progress broadcast and acquire and present caption content concurrent with AV program content.

The time for acquisition and presentation of captions (if present at that moment) should be on the order of the time for acquisition and presentation of video and audio.

In order to properly identify and display captions and subtitles, the following metadata is signaled:

- Language: the dominant language of the closed caption text;

- Role: the purpose of the closed caption text; e.g., main, alternate, commentary;
- Display aspect ratio: the display aspect ratio assumed by the caption authoring in formatting the caption windows and contents;
- Easy reader: this metadata, when present, indicates that the closed caption text is tailored to the needs of beginning readers;
- Profile: this metadata indicates whether text or image profile is used;
- 3D support: this metadata, when present, indicates that the closed caption text is tailored for both 2D and 3D video.

Protocols and Management Layer

ATSC Link-Layer Protocol (ALP)

ALP corresponds to the data link layer in the OSI 7-layer model. ALP provides a path to deliver IP packets, link layer signaling packets, and MPEG-2 Transport Stream (TS) packets down to the RF layer and back, after reception. ALP also optimizes the proportion of useful data in the ATSC 3.0 Physical Layer by means of efficient encapsulation and overhead reduction mechanisms for IP or MPEG-2 TS transport. ALP provides extensible headroom for future use.

The following functions are contained in ALP:

- ALP packet format;
- IP header compression;
- Link layer signaling.

The link layer is the layer between the physical layer and the network layer. The link layer transports the data from the network layer to the physical layer at the sending side, and transports the data from the physical layer to the network layer at the receiving side, as shown in Figure 5. While Figure 5 shows two logical flows between the link layer and physical layer, implementations are likely to utilize a single connection. The purpose of the link layer is to abstract all input packet types into a single format for processing by the physical layer (RF), ensuring flexibility and future extensibility for as-yet-undefined input types. In addition, processing within the link layer ensures that the input data can be transmitted in an efficient manner, for example by providing options to compress redundant information in the headers of input packets.

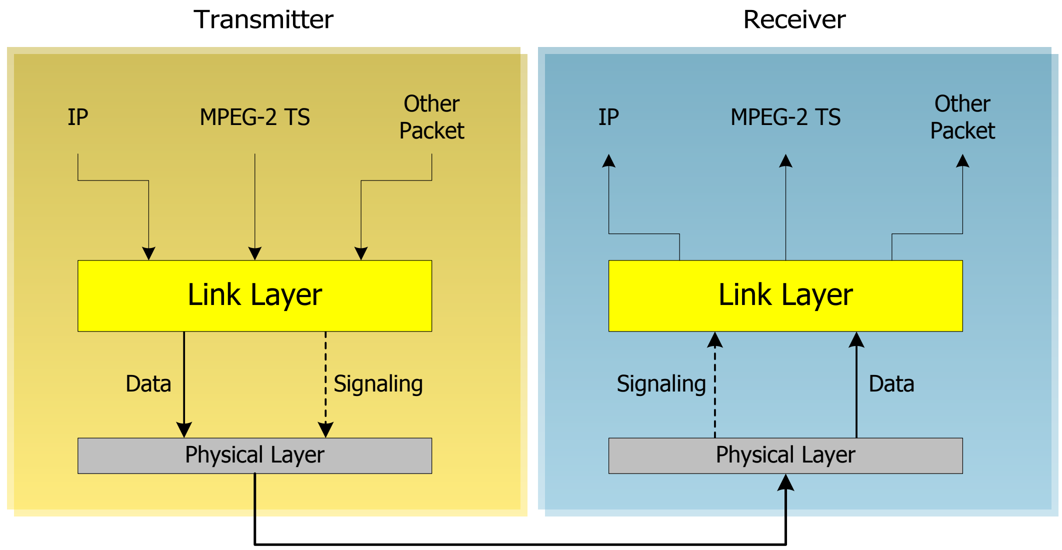

Figure 5: ATSC 3.0 link layer logical diagram

Source: ATSC Candidate Standard: Link-Layer Protocol A/330, Doc. S33-169r4, 24 May 2016

ALP allows encapsulation of any type of packet, including common ones such as IP packets and MPEG-2 TS packets. Using ALP, the physical layer need only process one single packet format, independent of the network layer protocol type (here we consider MPEG-2 TS packet as a kind of Network Layer Packet). Each Network Layer Packet, or input packet, is transformed into the payload of a generic ALP packet. Additionally, concatenation and segmentation can be performed in order to use the physical layer resources efficiently when the input packet sizes are particularly small or large. The term “IP packet” is used to refer to an IPv4 packet unless otherwise specifically noted.

When a Network Layer Packet is too large to process easily in the physical layer, it is divided into two or more segments. The link layer packet header includes protocol fields to perform segmentation on the sending side and reassembly on the receiving side. When the Network Layer Packet is segmented, each segment is encapsulated into an ALP packet and transmitted in the same order as its original position in the Network Layer Packet. Each ALP packet which includes a segment of a Network Layer Packet will be transported within the Physical (PHY) layer consecutively.

When the Network Layer Packet is small enough for the payload of a link layer packet to include several Network Layer Packets, the link layer packet header includes protocol fields to perform concatenation. Concatenation involves combining multiple small-sized packets into the payload of one ALP Packet. When the Network Layer Packets are concatenated, each Packet is concatenated into the payload of an ALP packet in the same order as the original input order. Also, each packet which constructs a payload of an ALP Packet is a whole packet, not a segment of packet.

Use of the ALP can result in significant reduction in overhead for transport of data on the physical layer. In ALP a specific format for signaling packets is provided to allow transportation of link layer signaling.

A functional architecture block diagram and interface is shown in Figure 6.

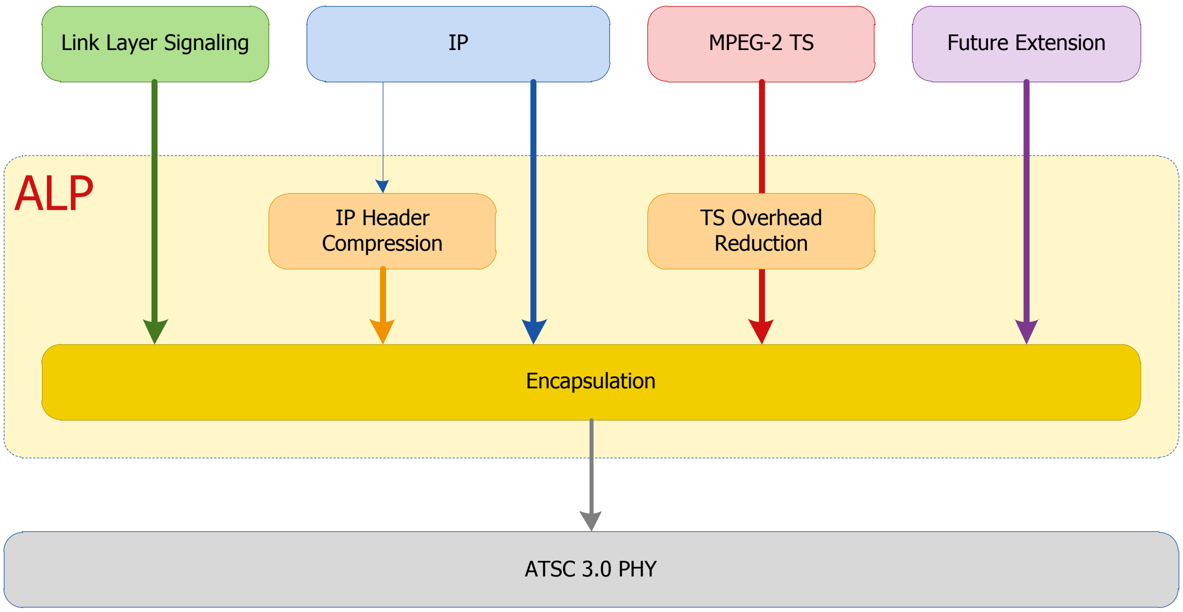

Figure 6: Block diagram of the architecture and interface of ALP
Source: ATSC Candidate Standard: Link-Layer Protocol A/330, Doc. S33-169r4, 24 May 2016

ALP takes in Network Layer Packets such as IPv4, IPv6 and MPEG-2 TS as input packets. Future extension indicates other packet types and protocols which are also possible to be input into ALP. ALP also specifies the format and signaling for any link layer signaling, including information about mapping specific IP packet streams to data pipes in the physical layer. ALP also incorporates mechanisms to improve the efficiency of transmission, via various header compression and deletion algorithms.

Signaling, Delivery, Synchronization, and Error Protection

A Broadcast Stream is an emitted signal on an RF Channel, which has a carrier frequency centered within a specified bandwidth (6 MHz for North America). A Physical Layer Pipe (PLP) corresponds to a portion of the RF channel. Each PLP has certain modulation and coding parameters and is identified by a PLP identifier (PLPID), which is unique within the Broadcast Stream it belongs to.

Each service is identified by two forms of service identifier: a compact form that is used in the Service List Table (SLT) and is unique only within the broadcast area, and a globally unique form that is used in the Service Layer Signaling (SLS) and the Electronic Service Guide (ESG). A ROUTE session is identified by a source IP address, destination IP address and destination port number.

In general, it is assumed that ATSC 3.0 services will be delivered by broadcast channel(s) and broadband channel(s) jointly leveraging the unidirectional one-to-many broadcast paradigm (either in fixed environment and/or mobile environment), and the bi-directional unicast paradigm in a broadband environment. The Service Announcement function enables ATSC 3.0 service providers to describe the ATSC 3.0 services that they make available. From a user's point of view, the Service Announcement function enables an on-screen Service Guide that can be seen as an entry point to discover ATSC 3.0 services and to select services. Service Announcement provides descriptions of the content offerings, and also may provide a filtering capability based on user preferences and content properties, such as the presence or absence of captioning, interactive enhancements, video formats (3D, SD, HD, UHD), audio formats (stereo, 5.1, immersive), content advisory ratings, genre, accessible audio tracks, and alternate languages, etc. In the case of scheduled services, Service Announcement also provides information about the date and time each offering is scheduled to be broadcast.

This layer of the system contains the technical mechanisms and procedures pertaining to service signaling and IP-based delivery of a variety of ATSC 3.0 services and contents to ATSC 3.0-capable receivers over broadcast, broadband and hybrid broadcast/broadband networks. The service signaling functionality defines the data formats and information components necessary to discover and acquire user services. The IP-based delivery functionality specifies two application transport protocols for the carriage of media content and service signaling data over broadcast and/or broadband networks to receivers. The delivery functionality also includes mechanisms for the synchronization of media components delivered on the same or different transport networks, and application-layer forward error correction methods that enable error-free reception and consumption of media streams or discrete file objects.

A conceptual model of the system showing the functionality found in all the layers is presented in Figure 7.

Two methods of broadcast service delivery are supported in ATSC 3.0. The method depicted on the left side of Figure 7 is based on MMT, ISO/IEC 23008-1 and uses Multi Media Transport Protocol (MMTP) to deliver Media Processing Units (MPU). The method shown in the center is based on MPEG-DASH, an adaptive streaming technique that packages and allows streaming of content over the internet from servers. It uses ROUTE protocol to deliver DASH Segments. Content not intended for rendering in real time as it is received, for example, a downloaded application, a file comprising continuous or discrete media and belonging to an app-based enhancement, or a file containing ESG or EA information, is also delivered by ROUTE. Signaling may be delivered over MMTP and/or ROUTE, while Bootstrap Signaling information is provided by the means of the SLT.

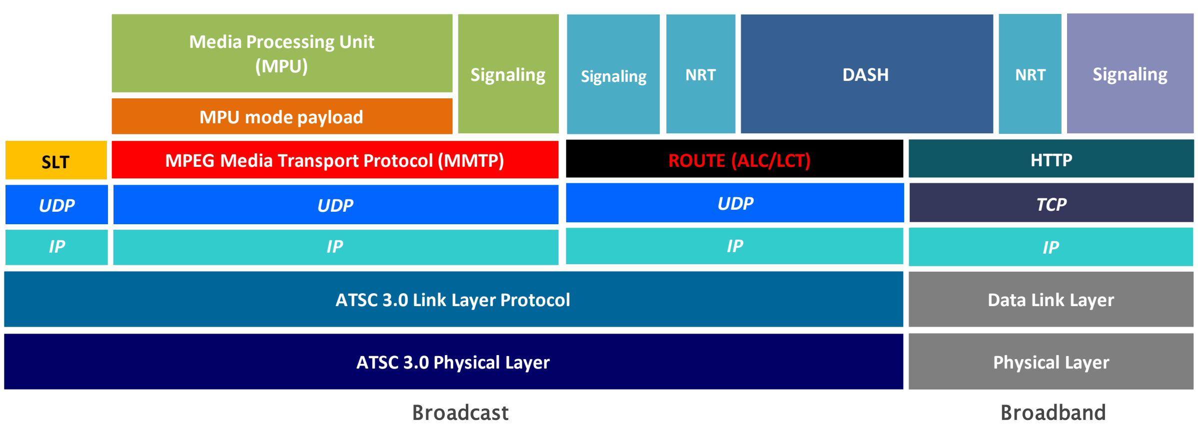

Figure 7: ATSC 3.0 receiver protocol stack

Source: ATSC Candidate Standard: Signaling, Delivery, Synchronization, and Error Protection A/331, Doc. S33-174r3, 21 June 2016

To support hybrid service delivery in which one or more program elements are delivered via the broadband path, the DASH-IF profile over HTTP/TCP/IP is used on the broadband side. Media files in the DASH-IF profile based on the ISO BMFF are used as the delivery, media encapsulation and synchronization format for both broadcast and broadband delivery.

ATSC 3.0 protocols included in the conceptual model provide support for system features including:

- Real-time streaming of broadcast media;
- Efficient and robust delivery of file-based objects;
- Support for fast service acquisition by receivers (fast channel change);
- Support for hybrid (broadcast/broadband) services;
- Highly efficient Forward Error Correction (FEC);
- Compatibility within the broadcast infrastructure with formats and delivery methods developed for (and in common use within) the internet;
- Support for Digital Rights Management (DRM), content encryption, and security;
- Support for service definitions in which all components of the service are delivered via the broadband path (note that acquisition of such services still requires access to the signaling delivered in the broadcast);
- Signaling to support state-of-the-art audio and video codecs;
- Non-real-time delivery of media content;
- Non-multiplexed delivery of service components (e.g., video and audio in separate streams);
- Support for adaptive streaming on broadband-delivered streaming content;
- Appropriate linkage to application-layer features such as ESG and the ATSC 3.0 Runtime Environment.

Service Announcement: Service Guide

Five basic types of ATSC 3.0 services are currently defined:

- Linear Audio/Video;
- Linear Audio-Only;
- App-Based;
- Electronic Service Guide;
- Emergency Alerting.

The Service Management layer primarily supports the means for service discovery and acquisition to enable different types of services, such as linear TV and/or HTML5 application service, to be carried by the underlying Delivery and Physical layers.

Service Signaling provides service discovery and description information, and comprises two functional components: Bootstrap Signaling via the SLT and SLS. These represent the information that is necessary to discover and acquire ATSC 3.0 services. The SLT enables the receiver to build a basic service list, and bootstrap the discovery of the SLS for each ATSC 3.0 service.

The SLT can enable very rapid acquisition of basic service information. The SLS enables the receiver to discover and access ATSC 3.0 services and their content components.

For ROUTE/DASH services delivered over broadcast, the SLS is carried by ROUTE/UDP/IP in one of the Layer Coding Transport (LCT) channels comprising a ROUTE session, at a suitable carousel rate to support fast channel join and switching. For MMTP/MPU streaming delivered over broadcast, the SLS is carried by MMTP Signaling Messages, at a suitable carousel rate to support fast channel join and switching. In broadband delivery, the SLS is carried over HTTP(S)/TCP/IP.

Interconnecting the Layers and Managing the System

The ATSC 3.0 system comprises a number of layers that must be connected to one another to construct a complete implementation. Two of the layers that must be interconnected are the transport layer and the physical layer. In addition, the physical layer is designed to be implemented partially at the studio or data source and partially at one or more transmitters. To enable the necessary interoperation of the layers and system segments, appropriate protocols are necessary so that equipment from multiple suppliers can be assembled into a working system. The two protocols that enable this interoperability are the ALPTP and the Studio-to-Transmitter Link Transport Protocol (STLTP), for carriage of data through specific portions of the system including the STL and transmitter(s). A Scheduler is also needed to manage operation of the physical layer subsystems and two protocols used by the Scheduler to receive high-level configuration instructions from a System Manager, and to provide real-time bit-rate control information to data sources sending content through the transport layer for emission by the physical layer.

The STL subsystem exists between the transport layer, which creates ALP packets, and the physical layer, which formats streams of ALP packets for transmission in particular PLPs in an emission configuration specified continuously in detail by a Scheduler.

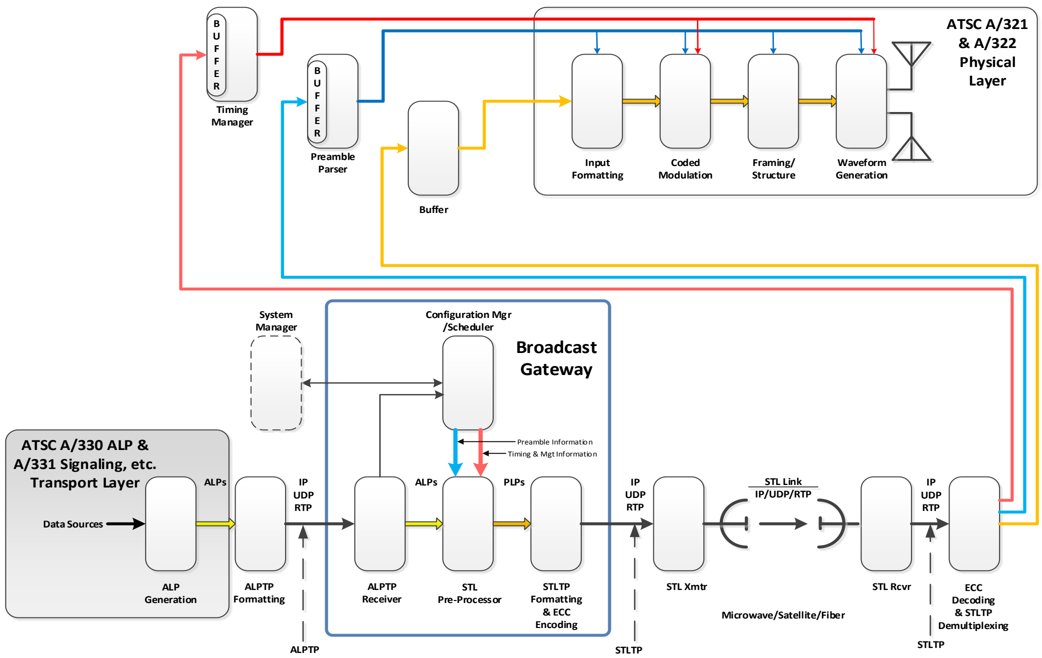

Figure 8: High Level Overview of System Configuration

Source: ATSC Working Draft: S32-Scheduler / Studio to Transmitter Link, Doc. S32-266r10, 6 August 2016

There is a one-to-one correspondence between individual streams of ALP packets and individual PLPs. To prepare ALP packets for transmission, the ALP packets are encapsulated in Base Band Packets (BBPs), which have defined sizes that are determined by a parameter related to the specific characteristics of the particular PLP(s) in which they will be carried. The sizes of the BBPs in a given stream are set to assure that the assigned capacity of the related PLP in a frame is filled by the BBPs derived from an associated ALP packet stream. ALP packets either are segmented or are concatenated so that they fill the allocated space in the BBPs, carrying them as completely as possible without overflowing the available space.

To manage the flow of data through the system, several buffers are required to hold data for purposes of time alignment of data emission.

Maintaining the one-to-one correspondence between particular ALP packet streams and their assigned PLPs through the system requires a method for identifying both the ALP and PLP data streams. Since the ATSC 3.0 system works in an IP environment, IP/User Datagram Protocol (UDP)/Real Time Protocol (RTP)/Multicast stacks are used in both of the ALPTP and STLTP structures.

To manage all of the characteristics of the emission and to coordinate all of the elements of the Physical Layer subsystem with respect to their parameter settings and times of operation, a Scheduler function is included in the Broadcast Gateway, which is a device that performs multiple functions that are part of assembling, configuring and scheduling all of the services that make up the overall data broadcast from a station. The Scheduler controls the generation of BBPs destined for each PLP, and creates the signaling data transmitted in the Preamble as well as signaling data that controls creation of Bootstrap signals by the transmitter(s) and the timing of their emission. To perform its functions, the Scheduler communicates with a System

Manager to receive instructions and with the source(s) of the ALP packets both to receive necessary information and to control the rate(s) of their data delivery.

One of the principal functions of the Scheduler is to generate Preamble data for the transmitter(s) that it controls. The Preamble generation function is assigned to a Preamble Generator, which is part of the Broadcast Gateway. The Preamble Generator outputs the data to be transmitted to receivers to allow their configurations to match the processes and parameters that will be used in transmission. As the transmitter(s) process the Preamble data for emission, it also will be used to set up the Input Formatting, Coded Modulation, Framing/Structure, and Waveform Generation so that the emitted waveform will match what receivers will be instructed by the Preamble to receive.

Similarly, the Scheduler must control the generation and emission of Bootstrap waveforms by the transmitter(s). To accomplish this, a data structure similar to the Preamble is required. A Timing and Management Data Generator is included in the Broadcast Gateway to provide this function under control of the Scheduler.

BBP data are carried across the STL as an IP/UDP/Multicast stream for each PLP. These streams are multiplexed into a single IP/UDP/RTP/Multicast stream for each broadcast emission to enable reliable delivery to the transmitter(s) of correctly identified and ordered BBPs.

The Studio to Transmitter Link (STL) interface is typically located between the baseband framer and the FEC block. There only needs to be one Scheduler and one Base Band Framer per RF emission. Multiplexing of multiple services among stations on one RF emission can be accommodated by the Scheduler.

Broadcasters have a need to send studio-generated data to their transmitters. Usually those transmitters are not co-located at the studio. An STL interface between the Gateway, located at the studio, and the transmitter(s) known as the STLTP, is needed for interoperability between these devices. That interface is required to:

- Support IP/User Datagram Protocol (IP/UDP) IPv4 and addressing;
- Encapsulate data for the link;
- Provide a synchronization method to ATSC time for data and control;
- Provide signaling of the transmitter timing synchronization for data and control;
- Have defined maximum latency, so as to allow the emission time to be correct;
- Allow for redundancy.

The physical STL may operate on any fiber, satellite or microwave connectivity. IP must be supported on all link types. All STL TP IP packets are tunneled into a single IP address and port. The base band packets are on separate ports.

Transmission configuration aspects are controlled from one entity called the System Manager. This entity provides configuration parameters for the various system functions; for example, video encoders to operate at fixed PQ or constant bit rate, and to define the physical layer configuration for the Scheduler.

The inputs to the Scheduler include ALP packets with their associated delivery metadata. The System Manager defines system configuration, including the number and configuration of PLPs, and their constraints such as the maximum capacity assignable to each PLP. These constraints are fed to the Scheduler.

The output of the Scheduler defines specifics of the baseband framing of the data packets. The input combination of data and delivery metadata is converted into a description of physical layer configuration that controls which data is sent at which times via specific physical layer resources.

The System Manager controls configurations of the transmission chain. It controls the physical layer configuration with respect to the number and configuration of each PLP, the Services supplied on those PLPs, and the delivery sessions that support the Services that run in the PLPs. There could be a pre-determined schedule for service operation and the System Manager can be responsible for implementing it.

Figure 9 below shows a conceptual block diagram of the Broadcast Gateway with its associated interfaces. The Configuration Interface allows provision of aspects such as PLP definitions. The Studio Interface delivers content and signaling, subject to delivery metadata. The Scheduler can communicate control upstream on the Studio Interface via ALP TP. The SFN Interface communicates a complete description of a physical layer instance on a frame by frame basis to the transmitter or transmitters if an SFN is used.

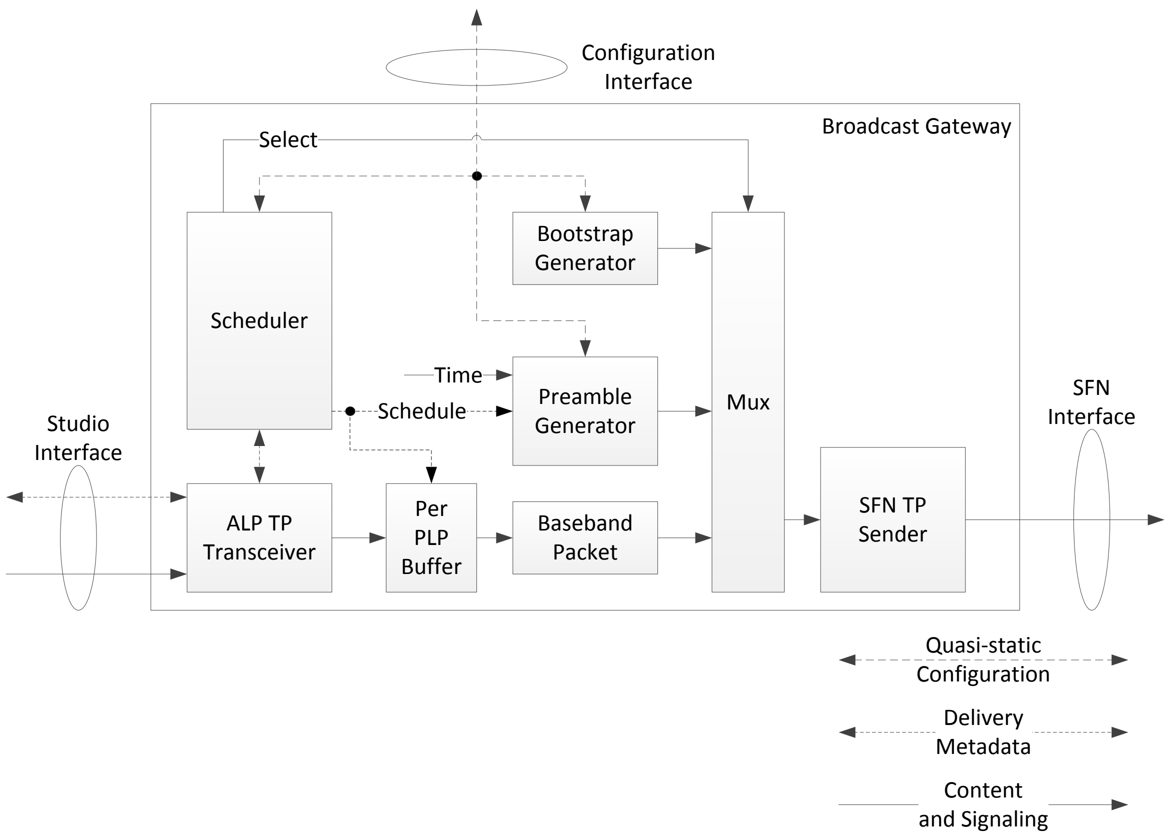

Figure 9: A broadcast gateway conceptual diagram

Source: ATSC Working Draft: S32-Scheduler / Studio to Transmitter Link, Doc. S32-266r10, 6 August 2016

This solution is subject to both the configuration and control parameters and the aggregate spectrum available. The Scheduler manages buffer fullness throughout the transmitter chain based on maximum delay of the network, maximum sizes of subframes/frames and using STL channel bandwidth allocation to IP port streams, which also requires determination of packet ordering in the STL. The Scheduler must be aware of timing throughout the transmitter chain and determine bootstrap emission time, and create timed control data for all transmitters.

The Scheduler defines physical layer frame lengths, subframe sizes, and bootstrap emission time, assigns frame identifiers based on bootstrap emission time, determines waveform configurations for frames, creates preambles according to physical frame configuration, manages baseband packet creation and ordering, and manages packet identification.

For Variable Bit Rate (VBR) encoding, there may be two feedback loops that the Scheduler is a part of: a short-time-frame feedback loop which controls the video/audio encoded bit rates on a per-physical-layer frame basis, and a slower configuration (control) loop. The slower loop may also contain data for services that are subject to quasi static rate control and that adapt more slowly, such as Non Real Time (NRT). This class of service may have entry control to utilize as much bandwidth as is opportunistically available. There may also be services that have static assigned rates and are not subject to dynamic management, other than turning the service on or off.

Physical Layer

The ATSC physical layer protocol is intended to offer far more flexibility, robustness and efficient operations than the ATSC A/53 (ATSC 1.0) standard, and as a result it is non-backwards-compatible with ATSC 1.0. This physical layer allows broadcasters to choose from among a wide variety of physical layer parameters for personalized broadcaster performance that can satisfy many different broadcaster needs. There is the capability to have high-capacity/low-robustness and low-capacity/high-robustness modes in the same emission. Technologies can be selected for special use cases like SFNs, Multiple Input Multiple Output (MIMO) channel operation, channel bonding and more, well beyond a single transmitting tower. There is a large range of selections for robustness including, but not limited to, a wide range of guard interval lengths, FEC code lengths and code rates.

Significant flexibility comes from a signaling structure that allows the physical layer to change technologies and evolve over time, while maintaining support of other ATSC systems. The starting point of this change is a physical layer offering highly spectral efficient operation with strong robustness across many different modes of operation.

The ATSC physical layer protocol is intended to offer the flexibility to choose among many different operating modes, depending on desired robustness/efficiency tradeoffs. It is built on the foundation of Coded Orthogonal Frequency Division Multiplex (COFDM) modulation (a method of encoding digital data on multiple carrier frequencies) with a suite of Low-Density Parity Check (LDPC) FEC codes, of which there are 2 code lengths and 12 code rates defined. There are three basic modes of multiplexing: time, layered and frequency, along with three frame types of Single Input Single Output (SISO), Multiple Input Single Output (MISO) and MIMO. Guard intervals are adjustable, with 12 selectable guard interval lengths to offer long echo protection. Channel estimation can be done with 16 scattered pilot patterns along with continual pilot patterns. Three Fast Fourier Transform (FFT) sizes (8K, 16K and 32K) offer a choice of Doppler protection, depending on the anticipated device mobility.

Supported bit rates in a 6MHz channel range from less than 1Mbps in the low-capacity most-robust mode, up to over 57Mbps when using the highest-capacity parameters. Data are carried in PLPs, which are data structures that can be configured for a wide range of trade-offs between signal robustness and channel capacity utilization for a given data payload. Multiple PLPs can be used to carry different streams of data, all of which are required to assemble a complete delivered service.

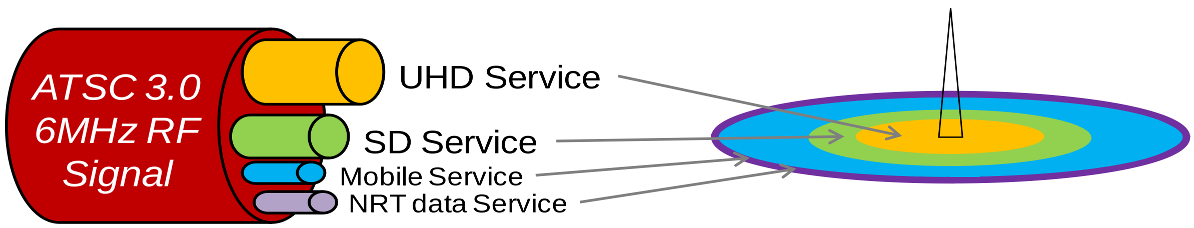

Figure 10: Example of multiple services carried within individual Physical Layer Pipes
Source: Hitachi-Comark

In addition, data streams required to assemble multiple delivered services can share PLPs if those data streams are to be carried with the same levels of robustness. Combinations of data streams necessary to assemble a particular delivered service to a receiver are limited to carriage on a maximum of 4 PLPs; however, a transmission channel can support up to 64 PLPs. These capabilities enable scenarios such as robust audio, video, enhanced video, and application data each being sent on an individual PLP at different robustness levels. A common source of channel interference is time-based burst noise. The use of a time interleaver, which can be configured for intra-subframe interleaving up to 200msec and larger depths in inter-subframe interleaving for low-bit rate streams, provides a method to reduce the interference at the receiving device. Frequency interleaving can be used throughout the channel bandwidth on a per symbol basis to separate burst errors in frequency domain.

The transmitted signal is organized into frames. A frame consists of a combination of three basic components, as shown in Figure 11:

- One bootstrap, located at the beginning of each frame. The exact time period from the start of a bootstrap to the start of the next bootstrap that matches the same major and minor bootstrap versions are an integer multiple of the sample time of the baseband sampling rate indicated by the first bootstrap.
- One preamble, located immediately following the bootstrap. The preamble will contain L1 control signaling applicable to the remainder of the frame.
- One or more subframes, located immediately following the preamble. If multiple subframes are present in a frame, then those subframes will be concatenated together in time as shown in Figure 11.

Each frame begins with a preamble which contains the signaling required for Layer 1.

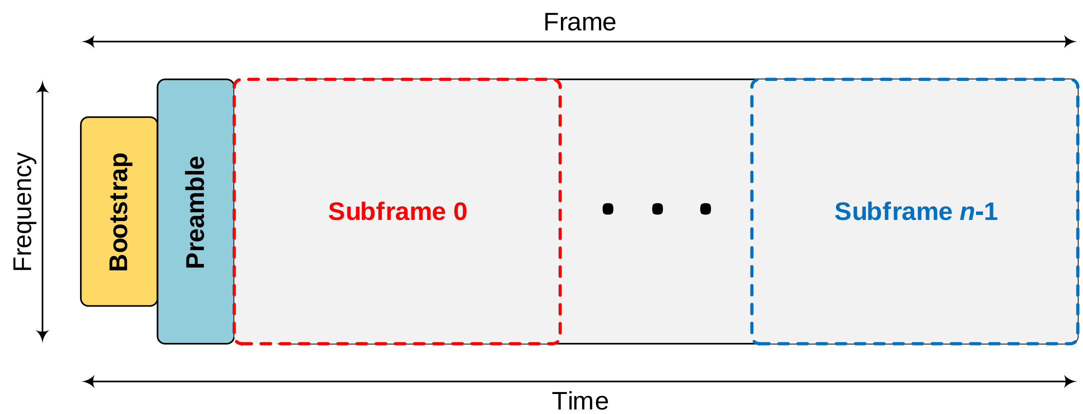

Figure 11: Frame structure

Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

The purpose of the physical layer is to offer a wide range of tools for broadcasters to choose the operating mode(s) that best fits their needs and targeted devices. This toolbox of technology is expected to grow over time, and the ability to upgrade or swap out new technology is enabled with the extensive and extensible signaling in the Preamble, which contains critical signaling and is found at the beginning of each frame or group of subframes.

The bootstrap is a short and robust signal that provides the receiver with information about signal discovery, channel estimation, receiver wake up for Emergency Alert System (EAS) alerts, preamble parameters and the version of ATSC that is being transmitted.

Broadcasters anticipate providing multiple wireless-based services, in addition to just broadcast television, in the future. Such services may be multiplexed together within a single RF channel. As a result, there exists a need to indicate, at a low level, the type or form of a signal that is being transmitted during a particular time period, so that a receiver can discover and identify the signal, which in turn indicates how to receive the services that are available via that signal.

To enable such discovery, the bootstrap signal is used. This comparatively short signal precedes, in time, a longer transmitted signal that carries some form of data. New signal types, at least some of which have likely not yet even been conceived, could also be provided by a broadcaster, and identified within a transmitted waveform through the use of a bootstrap signal associated with each particular multiplexed signal. Some future signal types indicated by a particular bootstrap signal may even be outside the scope of the ATSC.

The bootstrap provides a universal entry point into a broadcast waveform. The bootstrap employs a fixed configuration (e.g., sampling rate, signal bandwidth, subcarrier spacing, time-domain structure) known to all receiver devices, and carries information to enable processing and decoding the signal associated with a detected bootstrap. This capability ensures that broadcast spectrum can be adapted to carry new signal types that are preceded by the universal entry point provided by the bootstrap, for public interest to continue to be served in the future.

The bootstrap has been designed to be a very robust signal and detectable even at low signal levels. As a result of this robust encoding, individual signaling bits within the bootstrap are comparatively expensive in terms of the physical resources that they occupy for transmission. Hence, the bootstrap is generally intended to signal only the minimum amount of information required for system discovery (i.e., identification of the associated signal) and for initial decoding of the following signal.

The bootstrap provides a universal entry point into a digital transmission signal. It employs a fixed configuration (e.g., sampling rate, signal bandwidth, subcarrier spacing, and time-domain structure) known to all receiver devices.

The bootstrap consists of a number of symbols, beginning with a synchronization symbol positioned at the start of each frame period to enable signal discovery, coarse synchronization, frequency offset estimation, and initial channel estimation. The remainder of the bootstrap contains sufficient control signaling to permit the reception and decoding of the remainder of the frame to begin.

ATSC 3.0 has two concepts at its core: flexibility and efficiency. For flexibility, the number of modulation and coding combinations offers a breadth of choice for operating point(s) that have not been available in any previous broadcasting standard. For efficiency, the newest Low-Density Parity Check (LDPC) forward error correcting options with the adoption of non-uniform-constellations has brought the operation of the Bit Interleaved Coded Modulation (BICM) closer to the theoretical Shannon Limit: offering over 1dB of gain compared to the use of uniform constellations using the same operating parameters.

Multiple multiplexing methods give options to the broadcaster to configure the transmission chain like never before.

Signal Formatting and Coding

A block diagram of the main data flow for the total transmitter system architecture is shown in Figure 12. The system architecture consists of four main parts: Input Formatting, BICM, Framing and Interleaving, and Waveform Generation. For simplicity, control and signaling information flow is not shown in this diagram.

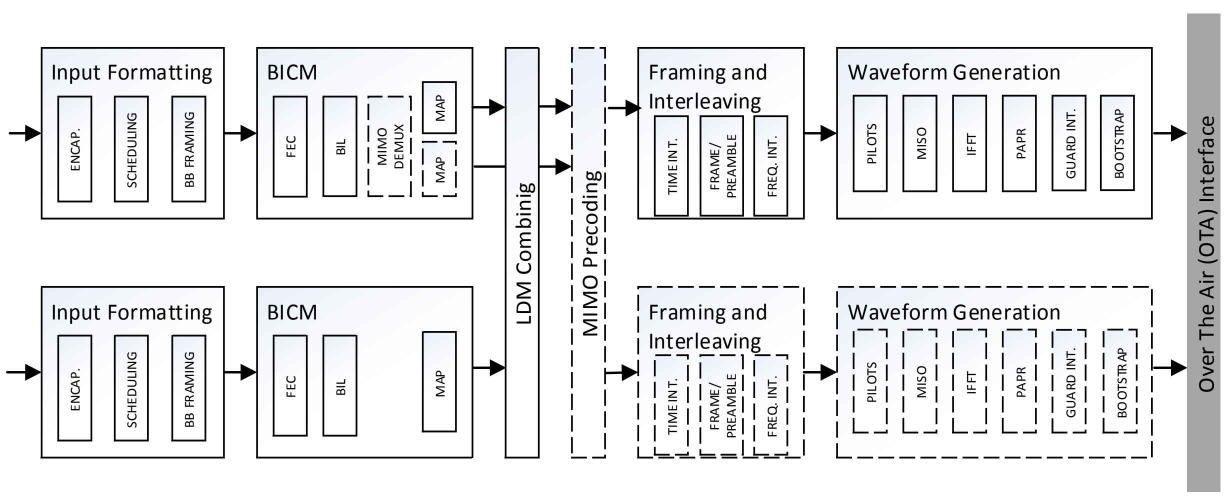

Figure 12: Block diagram of the system architecture for one RF channel

Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

Not all blocks are used in each configuration. In Figure 12 the solid lines show blocks common to Layered Division Multiplexing (LDM) and MIMO, dotted lines show blocks specific to LDM (MIMO blocks are not used), and dashed lines show blocks specific to MIMO (LDM blocks are not used).

The input formatting consists of three blocks: encapsulation and compression of data, baseband framing and the scheduler. This is shown in Figure 13. The dotted line represents the flow of control information, while the solid lines represent the flow of data.

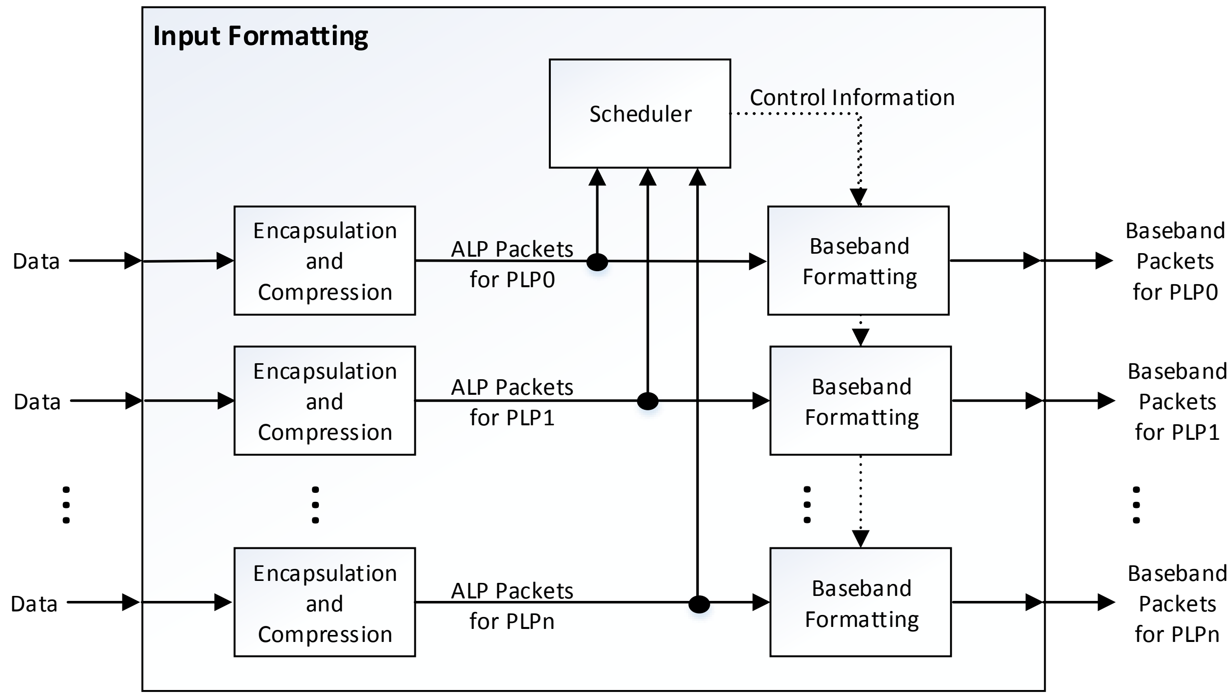

Figure 13: Block diagram of input formatting

Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

The baseband formatting block creates one or more PLPs as directed by the Scheduler. At the output of the baseband formatting block, each PLP consists of a stream of Baseband Packets, and there is exactly one Baseband Packet per defined FEC Frame.

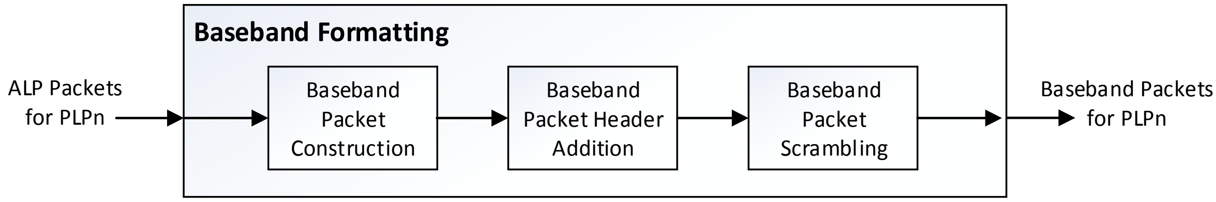

Figure 14: Block diagram of baseband formatting
Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

A Baseband Packet will consist of a header and a payload containing ALP packets, shown in Figure 14. ALP provides a path to deliver IP packets, link layer signaling packets, or MPEG-2 TS packets down to the RF layer and back, after reception.

Data padding, if required, will be added to the Baseband Packet Header. Baseband Packets have fixed length, with the length determined by the outer code type, inner code rate and code length chosen for the target PLP.

The Baseband Packet Header is composed of up to three parts. The first part is called the Base Field and appears in every packet. The second part is called the Optional Field. The third part is called the Extension Field. The order of the fields is Base, Optional and Extension. The Optional Field may be used to provide signaling regarding the following Extension Field. When Extension Fields are used, the Optional Field will always be present.

Next comes the BICM block, consisting of three parts: The FEC, the Bit Interleaver and the Mapper. The BICM block operates separately on each PLP.

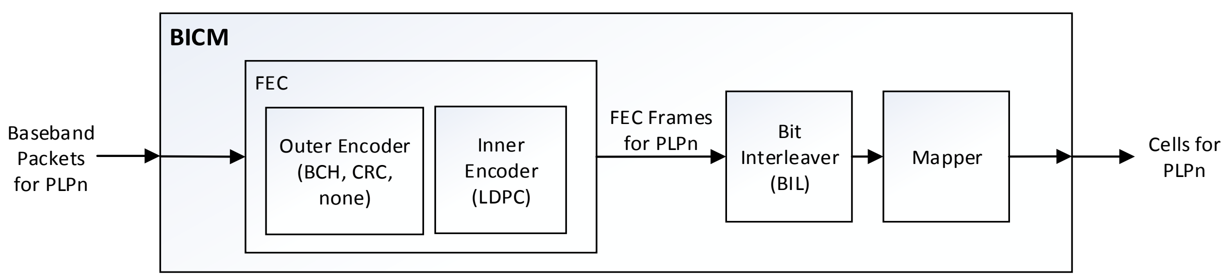

Figure 15: Block diagram of BICM
Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

The input to the FEC portion of the BICM is a Baseband Packet and the output is a FEC Frame. It should be noted that the size of the input Baseband Packet depends on the inner code rate and length and outer code type. The size of the FEC Frame depends on the code length only.

The bit interleaver block takes a FEC Frame as input. The purpose of interleaving is to provide some immunity to time-based noise busts that interfere with the received signal. The output of the bit interleaver block is a bit-interleaved FEC Frame. The size of the FEC Frame does not change after the bit interleaving operation. The bit interleaver block consists of a parity interleaver, followed by a group-wise interleaver, followed by a block interleaver. A block diagram showing the bit interleaver internal structure is shown in Figure 16.

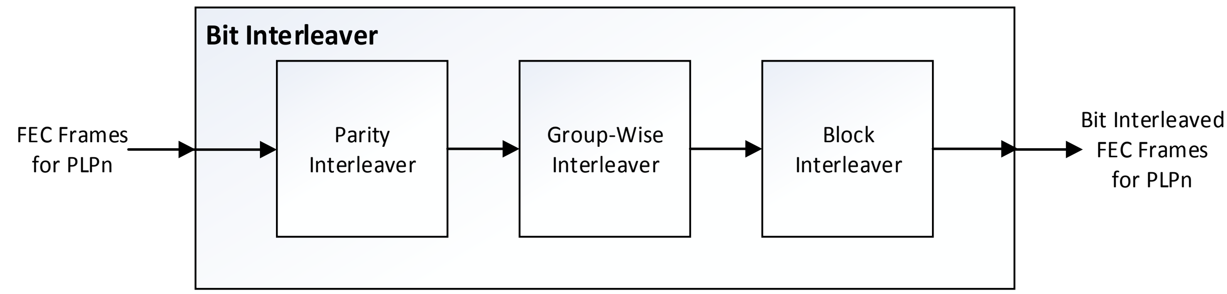

Figure 16: Bit interleaver structure

Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

A key concept in the Input Formatting and BICM blocks is the PLP, which is a stream of data encoded with a specific modulation, coding rate and length. For each PLP a separate Input Formatting and BICM block is used. It is noted that after the Framing and Interleaving block there is only one stream of data, as the PLPs have been multiplexed onto Orthogonal Frequency Division Multiplexing (OFDM) symbols and then arranged in frames.

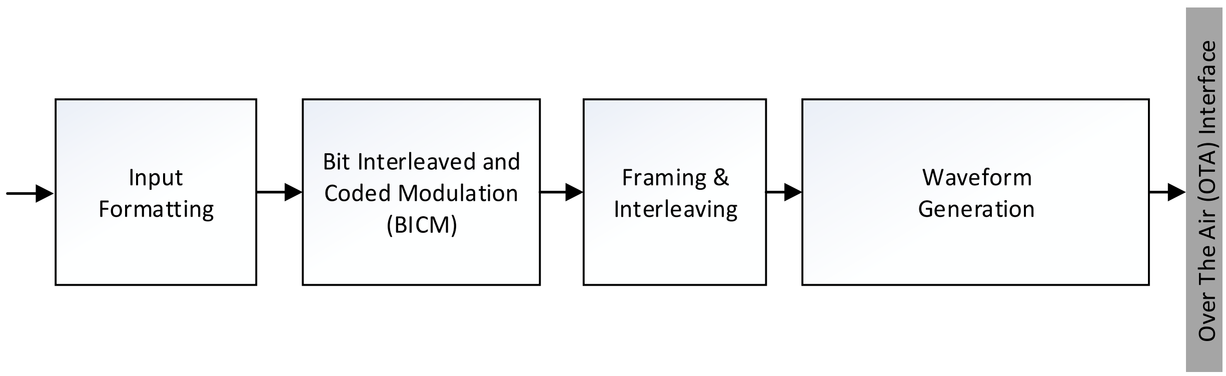

Figure 17: Block diagram (simplified) of a single PLP system architecture

Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

Although there are multiple methods of multiplexing the input data supported in this standard, two are described very briefly: Time Domain Multiplexing (TDM) and Layered Division Multiplexing (LDM). The system architecture block diagrams for these two methods show how the overall system architecture diagram can be simplified for specific configurations.

In the TDM system architecture there are four main blocks: Input Formatting, BICM, Framing and Interleaving, and Waveform Generation. Input data is formatted in the Input Formatting block, and FEC applied and mapped to constellations in the BICM block. Interleaving, both time and frequency, and frame creation are done in the Framing and Interleaving block. It is in this block that the TDM of the multiple PLPs is done. Finally, the output waveform is created in the Waveform Generation block. The TDM system architecture can be realized using the simplified block diagram shown in Figure 17.

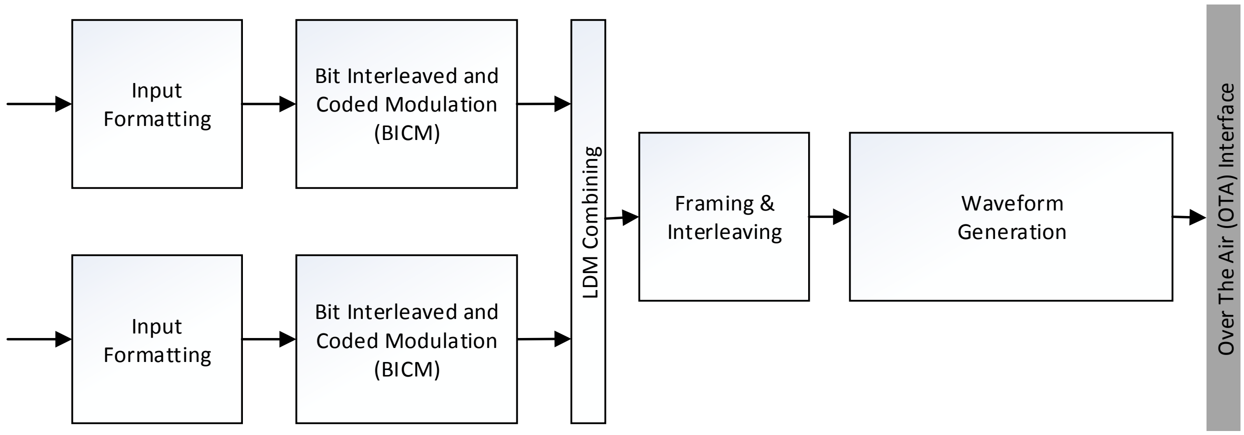

Figure 18: Block diagram (simplified) of the LDM system architecture
Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

LDM is a multiplexing scheme where multiple RF signals are layered on top of one another. A two-layer system has a core layer, which is more robust, and an enhanced layer, which is less robust. The enhanced layer is “injected” between -3 and -10dB relative to the core layer. LDM offers the broadcaster the advantage of being able to transmit the same content stream at two different levels of robustness, such that the receiver can fall back to the more robust signal if signal levels are not sufficient.

In the LDM system architecture, in addition to the four blocks that have already been shown in the TDM system, there is an additional block, LDM Combining. After combining the data from each layer, the data passes through the Framing and Interleaving block followed by the Waveform Generation block.

This standard also offers the option to use multiple RF channels through channel bonding, shown graphically in Figure 18. Compared to the TDM architecture, at the transmitter side there is an additional block, Stream Partitioning. The high data rate input stream is partitioned in this block into two separate streams, each passing through a BICM, Framing and Interleaving and Waveform Generation block. Each stream is output onto a separate RF channel. At the receiver side, the outputs of the two RF channels are then combined to achieve greater data rates than can be achieved in one RF channel alone.

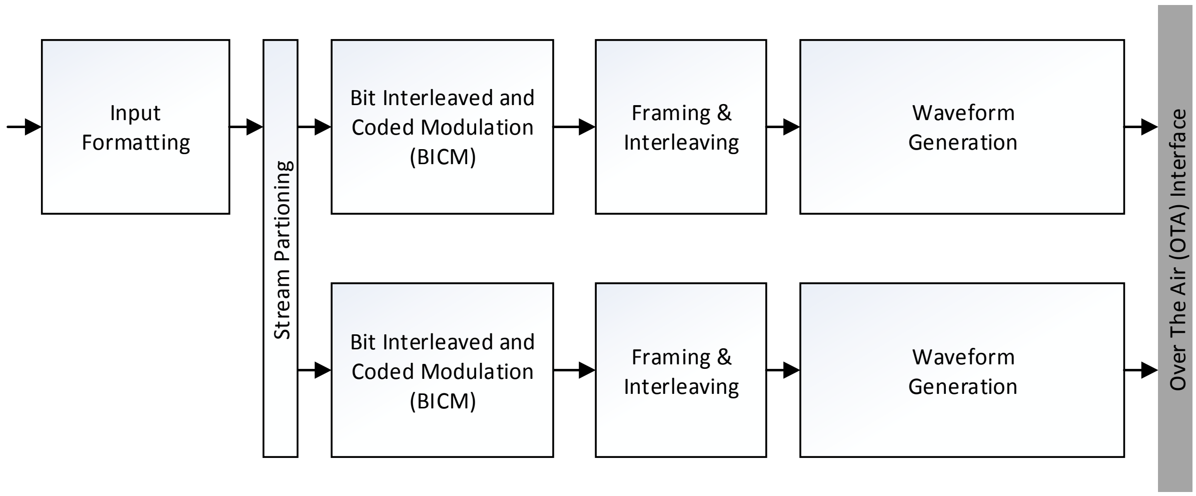

Figure 19: Block diagram (simplified) of a channel bonded system
Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

The Framing and Interleaving block consists of three parts: time interleaving, framing and frequency interleaving. The input to the time interleaving and framing blocks consists of one or more PLPs; however, the output of the framing block is OFDM symbols, either Preamble or Data, which are arranged in the order in which they appear in the final frame. The frequency interleaver operates on OFDM symbols. A block diagram of the framing and interleaving is shown in Figure 20.

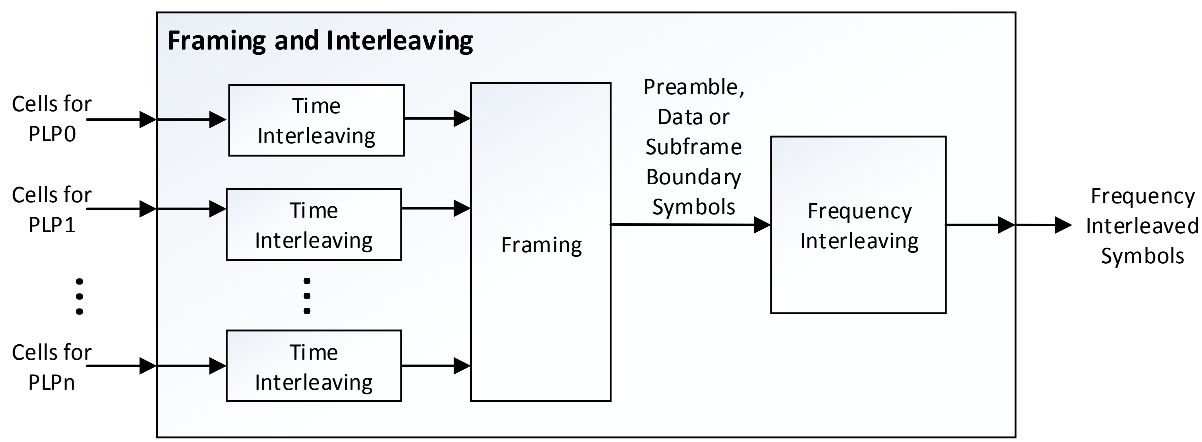

Figure 20: Block diagram of framing and interleaving
Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

The input to the Time Interleaving block is a stream of cells output from the Mapper block, and the output of the Time Interleaving block is a stream of time-interleaved cells.

Each PLP can be configured with one of the following time interleaver modes: no time interleaving, Convolutional Time Interleaver (CTI) mode, or Hybrid Time Interleaver (HTI) mode. The time interleaver mode for a PLP is indicated by the L1-Detail signalling field.

The Framing block takes inputs from one or more physical layer pipes in the form of data cells, and outputs frame symbols. Frame symbols represent a set of frequency domain content prior to optional frequency interleaving, followed by pilot insertion and then conversion to a time domain OFDM symbol via an Inverse FFT (IFFT) and guard interval insertion.

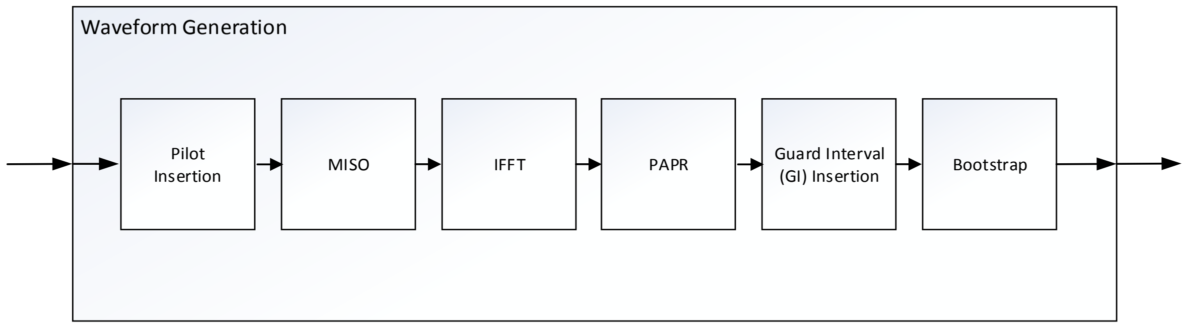

Figure 21: Block diagram of waveform generation

Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

The waveform generation section consists of pilot insertion followed by optional MISO pre-distortion. The resulting signal is passed through an IFFT, which is a mathematical model that converts items in the time or space domain into the frequency domain and vice versa. The math required is identical either way. In the case of a transmitter by convention it is known as an IFFT because the value is taken from the frequency domain (vector value of constellation) to the time domain (thousands of carriers in a single symbol). In a receiver, the opposite happens utilizing an FFT. The symbol value passes through the same mathematical construct and out pops the constellation vector value, and thus the symbol represented. As such, the FFT math translated the time domain value back to a frequency domain value.

So in practice how does this work? A particular symbol has a phase and amplitude component that has a value made up of a huge number of underlying frequencies of different amplitudes. When all these frequencies are added together, they make up the vector value. If a practical number of those frequency components could be derived mathematically in a relatively short time, they could each be used to modulate an individual carrier. The more carriers, the more accurately the signal could be represented (with less noise). However, a single transmitter must transmit all these carriers at once over the whole channel, so each takes a finite amount of the available power. Therefore, the more carriers, the less power available to each carrier. So it's a tradeoff.

After the IFFT, optional peak-to-average-power reduction techniques may be applied, followed by guard interval insertion. Finally the bootstrap signal is prefixed to the beginning of each frame.

Various cells within the OFDM frame are modulated with reference information whose transmitted value is known to the receiver. Cells containing reference information may be transmitted at a boosted power level. These cells are called scattered, continual, edge, Preamble or subframe boundary pilots. The value of the pilot information is derived from a reference sequence, which is a series of values, one for each transmitted carrier on any given symbol. The pilots can be used for synchronization, channel estimation, transmission mode identification and phase noise estimation, among other uses.

The MISO mode of operation uses a Transmit Diversity Code Filter Set, which is a pre-distortion technique that artificially pre-distorts signals from multiple transmitters in a Single Frequency Network in order to minimize potential destructive interference. The use of MISO is signaled on a per-subframe basis.

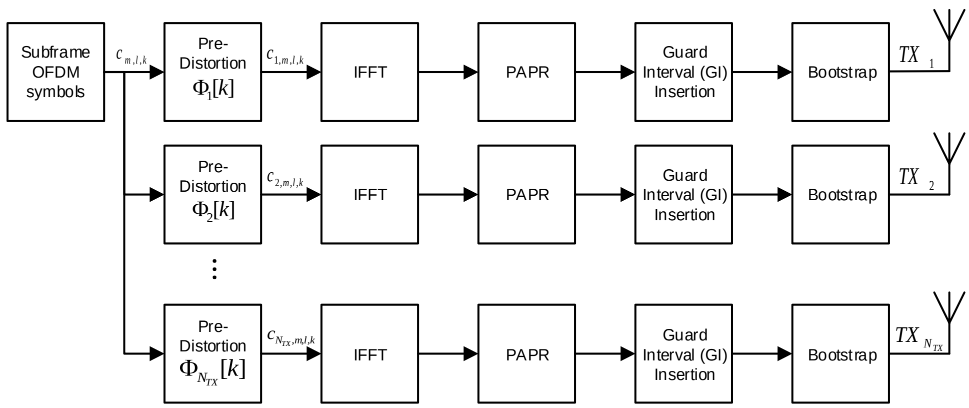

Figure 22: Block diagram showing example MISO transmission
Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

In order to reduce the Peak-to-Average Power Ratio (PAPR) of the output OFDM signal, modifications to the transmitted OFDM signal with tone reservation and Active Constellation Extension (ACE) may be used. None, one or both techniques may be used. Guard interval insertion is applied after the PAPR reduction.

Summary: ATSC 3.0 Standard

This ATSC 3.0 System Overview, while lengthy, is a consolidated and abbreviated summation of the content found in the primary ATSC 3.0 standards documents. These documents were written and published by the many contributors engaged in the development of the ATSC 3.0 suite of standards. Each of the standards documents contains much more detail than presented in this overview.

It should also be noted that some elements of the system have been left out of this overview to keep the focus primarily on ATSC 3.0's broadcast functionality. Content not included in the System Overview includes material on audio and video watermarking, security, runtime environment, companion devices, service usage reporting, and the return channel.

Readers are encouraged to become more familiar with the ATSC 3.0 documents as the industry gets closer to deployment. ATSC Standards and Recommended Practices are subject to change. These standards documents can be found at ATSC.org.

PART 2: REAL WORLD IMPLEMENTATION

ATSC 3.0 Service Models

The System Overview describes the theory behind ATSC 3.0. This section of the Guide offers a view of the real word practical implementations of the system from a business and technical perspective. Before making system-related technical decisions, it is necessary to define the types of services the station intends to provide. These upfront business decisions made early in the transition process, regardless of when those services will be implemented, have the potential to save the station significant time and money in the future. The decision to add these services (e.g. mobile) can impact the basic transmission system, while other decisions will impact the upstream portion of the broadcast plant. They can also impact the addition of hosting services for content delivered via the internet connection.

Ultimately adding these services will present a significant impact on how content is created at the station and how external content providers, such as affiliated network or syndicated programmers, are integrated into the new system.

One of these services, the delivery of traditional and enhanced linear programming as well as application-based functions, is well enabled by ATSC 3.0. Enhanced linear programming can include a variety of different content components such as multiple video, audio and caption streams, that can be selected and synchronously combined for presentation at the receiver. Linear programming services can be enhanced by downloadable receiver applications such as interactive games or targeted ad insertion.

Application-based services are also possible, in which an application serves as a service's launching point, and the service is consumed from within the application. An example of an application-based service is an on-demand offering allowing viewers to play back and manage a library of content. This is accomplished by combining the system's linear features with on-line delivered content.

One of the primary drivers behind the development of ATSC 3.0 is broadcasters' desire to deliver new services that make them competitive with other media delivery systems as well as expanding their business opportunities. No single technological change could satisfy this goal. There are a wide variety of possible enhancements, when carefully chosen, that can institute more competitive offerings, many of which are described in this section.

The following is a summary of the key service model enhancements that are built into the ATSC 3.0 system.

Mobile

The ability to receive local HD content anywhere in the world with a mobile device (smartphone, tablet, etc.) is a growing use case. ATSC 3.0's mobile reception performance is expected to be significantly better than that of previous broadcast systems, and the new system is designed to enable broadcasters to reach these devices. The mobile reception performance is enhanced by greater data error protection as well as better Doppler reception characteristics, improving reception even in moving vehicles.

Mobile device screens might be small, but they can support resolutions up to and including UHD video, as well as multichannel audio.

Delivering that content over broadcast networks is, and will remain, the most efficient method of distributing high-demand (live and non-real time) content to mobile devices simultaneously. The system is designed to optimize receiver battery life. Integration with common functionality on mobile receivers (such as social media and geo-location) can also be an important feature.

Ultra HD

The system supports video resolutions beyond HD (e.g., 3840 x 2160 pixels) and other improvements to fixed and mobile devices. Accompanying this higher resolution is an immersive audio experience with sound localization available on headphones or loudspeakers.

Other improvements (independent of video resolution) can include, for example, 10 bit or higher depth, 4:2:0 or 4:2:2 or 4:4:4 Y'CbCr Chroma subsampling schemes, wider color gamut, higher frame rates, higher dynamic range video, and audio object coding. State of the art video codecs (e.g., HEVC) and audio codecs are supported.

ATSC 3.0 allows for dividing the screen into separate sections to share a variety of presentation modes for UHDTV applications. These modes can be presented in several ways: as the main broadcast stream; an alternate piece of the main stream serving as a panning focal point; or multi-screen views that can incorporate internet web, photo or video browsing on the main screen or on a second screen, or simultaneously on both. These aspects combine broadcast and broadband content to provide an immersive and synchronized environment.

Hybrid Services

Content can be aggregated and combined from a variety of inputs and delivered via dynamically changing distribution channels, for both broadcast and broadband. Integration of these distribution channels opens up a variety of interactive services for the user, and also provides a transition to new services.

This connection allows a return path for user interactivity (e.g., Video on Demand), audience data collection and connection to social networks. Independent interactive applications may supplement broadcast content (e.g., closed captioning in unique languages, virtual reality ports) or make broadcast content and services more flexible and useful. This supplemental content can be synchronized with the broadcast stream.

Apps can be downloaded to devices with hybrid broadcast/broadband connections. This can enable, for example, free trials of interactive services or downloading of software updates that deliver improved user experience performance. The ability to easily download software also helps broadcasters immediately integrate new services (e.g., social apps, third-party vendor content) via embedding active service components. These scripts allow dynamically changing presentations of content depending on device type, or the ability to discover, communicate, synchronize or forward content to other devices (e.g. smartphones, tablets, PCs).

A mix of real-time and non-real-time delivery of content via broadcast and broadband paths is used to optimize productivity of the hybrid infrastructure and balance data bandwidth demand.

Multi-View/Multi-Screen

The system can provide multiple views associated with the same program, displayed on a single or multiple screens, by which the user can view different aspects of the same program, or information related to aspects of the program.

A user may select one of the available mash-up presentation modes available on the TV, and subsequently the screen is subdivided into a number of subparts. For example, in one subpart, the web page of a local business can be displayed showing details pertinent to the current broadcast. In another subpart, photos from community internet sites can be displayed showing these items in use. Hearing-impaired users may also rely on third-party applications for closed captioning for any subpart.

Users may chat with each other during a TV program through text messages appearing on a portion of their TVs, or on other personal devices using companion internet delivery. Content intended for personal devices can be synchronized with the primary TV content, streamed over the internet and displayed with HD quality on the personal device. What is displayed on the primary TV or personal devices may be selected via the personal device's control. Sharing certain views with social groups can occur when the receiving device is internet-connected, as the ATSC 3.0 system includes a return channel.

Users may also experience a panoramic view of sports programs, where multiple views of a sporting event are transmitted over both broadcast and broadband channels. The TV can seamlessly integrate the multiple views into one panoramic image. The TV remote controls can be used to pan, zoom or select individual views, possibly at different camera angles. Users can, for example, also select an athlete, and have that athlete remain at the center of the viewing action.

All of this is accomplished with the broadcaster specifying content distribution, content resolution, data rate, quality of service, and possibly other parameters, over a combination of broadcast and broadband transmissions. Receivers can synchronously combine content from different sources to render a seamless service.

3D Content (Video)

ATSC 3.0 also allows broadcasters to transmit enhanced depth information through a next-generation 3D transmission format. There are a number of ways this can be accomplished:

- 3D TV content and depth information can be delivered over the terrestrial broadcasting network, but an all-terrestrial transmission will require a higher bandwidth.
- Alternatively, 3D TV content can be delivered via terrestrial broadcasting, while the depth information can be delivered over the internet or broadband network. This is a hybrid delivery mechanism requiring synchronization between components delivered via different networks.

In addition, the system allows even greater personalization by enabling users to control the 3D depth via the receiver's remote controls. Greater flexibility is achieved by being able to transmit information that allows receivers to display captions in front of 3D TV content. Receiver-generated graphics then can be correctly positioned in relation to the 3D content depth.

Enhanced and Immersive Audio

The system supports high-quality audio presentation with perceptually transparent quality, highly accurate sound localization, and very high sense of sound envelopment within a targeted listening area ("sweet spot").

The system enables a universal, flexible format that allows rendering of proper spatial imaging of audio content on any kind of loudspeaker setup (e.g., with misplaced speakers, with fewer speakers than the number of audio channels encoded in the program, with a small number of "soundbars," or when using headphones).

The user can also change the balance of the loudness of dialog or commentary and ambient or background sound individually to suit personal preference. The mix may be varied to suit the listening environment, to emphasize the audio element most important to the user, or to enhance the speech intelligibility.

Accessibility

Captioning and Video Description services allow multiple closed-caption, subtitle and assistive audio services from which the user can choose. Assistive audio services can include one or more video description tracks, and/or one or more alternate-language dubs, with or without the original dialog remaining audible, or ducked under. These text and audio services can be delivered via either the broadcast channel or the Internet, with synchronization maintained.

Text services are delivered as discrete program streams, so they can be independently accessed by receivers and flexibly processed by users (e.g. sent to local text-to-speech converters, routed to second-screen devices, captured and stored as "transcript" files, etc.).

Captioning is delivered appropriately for both 2D and 3D video programs, with adequate user control to set preferences for caption display in both formats. Additional personalization allows for a main audio service with a feature allowing users to increase the dialog's intelligibility by varying the dialog's loudness with respect to other soundtrack elements.

Program guide information is presented in a flexible format that can be rendered in forms other than traditional on-screen text display (e.g., synthesized speech, Braille, IR audio). This data can be delivered via broadcast or internet, and the system can support receivers that include alternate control methods (such as voice-activation).

Users are able to set their accessibility preferences on one receiver, and store them in the cloud so they can be accessed when using a different receiver. Optionally, receivers can store multiple accessibility preferences locally, so once settings are retrieved from the cloud for a particular user, the retrieval process need not be repeated upon subsequent requests by that user on that receiver.

Advanced Emergency Alerting

Public emergency alerts via robust ATSC 3.0 broadcasts deliver basic and critical information. A key function of the alerting system is the ability to wake up receivers. In addition to providing the basic alert information, the message can be supplemented by broadcast of rich media, as well as providing links to other sources of information. Power-saving features of the physical layer allow long-time standby operation of handheld/portable receivers, which can wake up and notify the user in case of acute emergencies. Critical emergency information is delivered with high reliability and low latency, independent of other content in an RF channel.

The system is also very useful in providing emergency preparation guidance and post-event information, which typically will have longer lead and availability times, using rich media such as animated weather maps or escape routes. In this way content is presented to the user for optional viewing. Localization filtering in the receiver tailors the information to those who will be affected. The system delivers links to both extended and redundant information by multiple means (e.g., broadcast and broadband), increasing the likelihood of reaching all who need emergency information.

Handheld devices normally operate as primary receivers of alerts, but when used in conjunction with a fixed primary screen, a handheld device can provide supplemental “second screen” alerting functions and information.

Personalization/Interactivity

Some personalization and interactivity examples have been previously cited; however the list below summarizes the many ways television viewing may be personalized:

- Access to alternate primary video views, such as:
    - camera angles;
    - zoom-in or follow-subject in UHD;
    - alternate resolutions (dynamically adjusted by viewing device, or manually adjusted to save power, for example);
    - closed captioning on/off, choose style, choose languages.
- Access to alternate primary audio feeds, such as:
    - other languages;
    - alternate mixes (e.g., background ambience attenuated for increased intelligibility of dialog);
    - voice-over scene description;
- Access to secondary, but related content, such as:
    - Extra information (e.g., player statistics, product information, in-depth news);
    - Alternate versions of the primary content (e.g., longer/shorter versions, bonus content, virtual tour);
    - User-generated content (e.g., tweets, chats, social video-conferencing);
    - Interactive content (e.g., games, voting/polling).
- Set advertising preferences
    - Set preferred products and ad delivery mechanisms;
    - Opt out of targeted advertising (perhaps in exchange for a subscription fee);
- Set interactivity preferences
    - Choose desired apps, suppress unwanted apps.

In a given session, for example, two viewers can seamlessly move through a video entertainment landscape by accessing alternate content, simultaneously or sequentially, using interactive applications. Once those viewers are ready, they can return to the primary content at the point in which they left. A single viewing session could occur seamlessly across devices, on multiple devices simultaneously, across channels, between linear and non-linear viewing, and across delivery mechanisms (e.g., broadcast and broadband). Users' preferences follow them throughout their viewing sessions.

Viewers can also store preferences in the cloud so those preferences can be uploaded by any viewing device. This allows different people to customize their experiences, including accessibility preferences on shared devices such as a living room or on television sets outside the user's domain. Users can also store preferences locally on the viewing device, so that the preferences do not need to be uploaded for each viewing session.

Alternate content required to fulfill viewer preferences can be carried within the broadcast stream or accessed through an alternate means such as the internet. Such content is provided by the programmer/broadcaster or by a third party, and it may be synchronized or non-synchronized with the primary content.

In some cases preferences are accommodated entirely on the device, while in other cases a two-way communication path is required, in which media is sent to the viewer by the broadcaster, and interactive input is received by the broadcaster from the viewer. Interactive input may be manually sent (e.g., by the user clicking on something) or automatically triggered (e.g., based on geo-location data in a mobile device).

User preferences can be overridden by the broadcaster, as necessary, in the event of an Emergency Alert.

Advertising/Monetization

To address current trends in advertiser and consumer desires, the system enables broadcasters to adopt new and advanced advertising models.

Targeted advertising can be available within the primary television content and the secondary content displayed on companion devices. A selection of ads can be delivered with the broadcast stream, pre-loaded into the viewing device, or accessed from the cloud. Triggering mechanisms are present in the content so that the time, duration, placement and type of ad are identified to the device. Based on the viewing circumstances, one of the ads available to the viewing device is selected for presentation.

Another method for targeting advertising is by regionalization or geographic location. Two methods of enabling this type of ad targeting are possible. The first depends on the receiving device having the ability to determine its location, and the second method relies on distributed transmission, where specific ads can be targeted to a specific transmission area.

The new system will allow broadcasters to adopt other forms of advertising messages beyond the 30-second advertising spot, both within the primary content and within secondary content. Interactive applications during the broadcast or in secondary content can be sold to advertisers. Viewers can play a game, vote in a contest, complete a purchase, etc. Advertisers can license a portion of screen real estate for ad insertion during specific programs.

Advertising-delivery measurement is possible, including gathering demographic information about the viewers that experienced the message (subject to viewer opt-in for such data collection). Measurement of the advertising effectiveness is also possible, including reporting duration of time viewers spent with the message, level of interaction, number of purchases, etc.

New forms of measurement are particularly important for mobile devices, where traditional ratings systems may not properly reflect mobile viewing. In a mobile application environment, a substantial amount of end-user viewing telemetry could be collected via the device's wireless IP connection. This promises not just estimated viewership, but actual recorded viewership, along with viewing time and program choice. This hard data can be useful to broadcasters in their relationships with advertisers.

Merging the Service Model with Technology - Trade-offs between services, robustness and bandwidth

As the reader can see from the prior sections of this paper, ATSC 3.0 provides the broadcaster with many new opportunities to provide services, as well as to reach viewers in venues that have not been served by the current ATSC system. Developing a business model for a station will require selection of the services which the station wants to provide, selecting the best delivery (emission) method to reach the targeted viewers, and then matching those goals against the limitations of bandwidth.

A major difference between ATSC 1.0 and ATSC 3.0 is that ATSC 3.0 does not have fixed 19.39 Mbit/s payload bitrate. ATSC 3.0 bitrates range from 1.5 Mbit/s with a C/N of approximately -5dB, up to 57 Mbit/s requiring a C/N of well above 30dB for a 6 MHz channel.

Since ATSC 3.0 has the ability to provide a variety of services through multiple PLPs, each with individually assigned emission characteristics, the station operators can design an overall emission model that best suites the desired business model.

The emission of each PLP can be selected by a number of variables that will determine the robustness as well as the payload capabilities of that particular delivery pipe. The sum total of the PLP's will determine what will fit in the station's 6MHz television channel.

The variables that determine the robustness and the payload capacity of a PLP include the modulation scheme which could be selected from the following: QPSK, 16QAM, 64QAM, 256QAM, 1024QAM and 4096 QAM. These waveforms may be generated in uniform or non-uniform constellations. A good guide is that the more complex the modulation scheme, the greater the payload capacity, but the level of robustness decreases since the C/N requirement gets higher.

Another variable is the LDPC FEC codes that are adjustable in steps of 1/15 from 2/15 to 13/15. Again, a general rule is the more applied FEC the greater the robustness. However, this decreases the payload capacity because of the additional error correction bits required.

Additional variables include the time duration of the guard interval and the length of the data frame. This list of variables driving the overall emission configuration represents the starting point but there are additional variables that must be factored into the process for optional configurations like LDM, MIMO and MISO, and if time and frequency multiplexing are employed.

Each PLP's emission type must be customized for the desired receive situation, but must also support the required bitrates of the encoded content. The content payload requirements will be determined by video resolution along with the presence of optional video enhancement features like HDR, WCG and High Frame Rate (HFR).

The following table, Figure 23, illustrates six operating models that were developed by a group of broadcast engineering experts using use case models that have been selected by broadcasters. The table illustrates the types and numbers of services, the target type of reception, the modulation parameters of each PLP and the payload capability of each PLP. The models are each configured to make full use of the 6 MHz channel.

Potential Configuration	Opportunity	Multiplex Capacity	Targeted Receivers	Service Assignments	Channel Loading (BCN On)	PLP Capacity (Mbps)
1	UHD from Single Stick TDM	Lots more HD/SD services space available	Fixed	I UHD + 1-2 HD or I UHD + 3-6 SD	PLP1: 32K FFT; 148 usec GI, 64QAM, 64800 LDPC, 11/15 FEC, 250 msec Frame	20.03
	Audio Services		(similar to today)		PLP2: 32K FFT; 148 usec GI, QPSK, 64800 LDPC, 5/15 FEC, 250 msec Frame	0.66
2	Multicast HD/SD From single stick TDM	5-12	Fixed	2-4 HD or 8-10 SD in a single video stat mux pool	PLP1: 32K FFT; 148 usec GI, 64QAM, 64800 LDPC, 11/15 FEC, 250 msec Frame	17.59
	Audio Services		(similar to today)		PLP2: 32K FFT; 148 usec GI, QPSK, 64800 LDPC, 5/15 FEC, 250 msec Frame	1.03
3	UHD + Mobile HD	2-5	Fixed	1 UHD (+ audio)	PLP1: 32K FFT; 148 usec GI, 256QAM, 64800 LDPC, 11/15 FEC, 250 msec Frame	20.80
			Mobile & Indoor	4 SD or qHD Mobile (+ audio)	PLP2: 8K FFT; 148 usec GI, 16QAM, 64800 LDPC, 5/15 FEC, 250 msec Frame	2.23
4	Multicast HD/SD Robust Core SFN	5-7	Fixed	2 HD in video stat mux pool	PLP1: 16 K FFT, 148 usec GI, 64QAM, 64800 LDPC, 11/15 FEC, 250 msec Frame	8.70
			+ Robust Services	+3-5 SD in video stat mux pool	PLP2: 16 K FFT, 148 usec GI, 16QAM, 64800 LDPC, 5/15 FEC, 250 msec Frame	2.97
	Audio Services			+ audio	PLP3: 16K FFT; 148 usec GI, QPSK, 64800 LDPC, 5/15 FEC, 250 msec Frame	0.66
5	Deep indoor HD Core + Mobile single Stick	6-8	Fixed + portable deep indoor receivers	2 HD in video stat mux pool (+ audio)	PLP1: 16 K FFT, 148 usec GI, 64QAM, 64800 LDPC, 7/15 FEC, 250 msec Frame	5.38
				4-6 SD in video stat mux pool (+ audio)	PLP2: 16 K FFT, 148 usec GI, QPSK, 64800 LDPC, 7/15 FEC, 250 msec Frame	2.13
				+audio	PLP3: 16K FFT; 148 usec GI, QPSK, 64800 LDPC, 5/15 FEC, 250 msec Frame	0.66
6	Deep indoor & Mobile SFN TDM	Roughly 5	Portable receivers indoor and High Speed Mobile	4-5 SD or qHD in a video stat mux pool (+ audio)	PLP: 8 K FFT, 222 usec GI, 16QAM, 64800 LDPC, 5/15 FEC, 250 msec Frame	5.74

Figure 23: Six Use-Cases Operating Models

Source: ATSC Proposed Standard: Physical Layer Protocol A/322, Doc. S32-230r56, 29 June 2016

Most station engineers would not be able to configure, much less optimize, a station's emission for its business plans without understanding all variables and rules to consider. A solution to this problem is a Configuration Manager program that is associated with the Systems Scheduler. It is envisioned that this application will contain the rules and calculating engine that takes input from the many system equipment blocks and provides an operational configuration to the Broadcast Gateway and the Exciter. It will let the person requesting a configuration know if a valid emission configuration is possible, and prohibit any that would violate the standard or the 6MHz channel capacity.

Understanding the System Building Blocks for ATSC 3.0

Here the Guide lays out what engineers need to understand when implementing ATSC 3.0, especially for knowing where the processes and functions reside. The following description is based on hardware/equipment models that are either already implemented or under development by equipment providers.

Early implementations of ATSC 3.0 will be rolled out with a major focus on the building blocks, from the encoding platform out to the antenna. Some of the system blocks will require wholesale change, while others might only require some upgrading in order to support the transition. Later, as the networks and program suppliers deliver content in UHD formats, and stations produce local UHD content, other parts of the station must be updated to support these new formats. For the purposes of this document's iteration, the focus is on the transmission blocks, beginning with the encoding system.

Encoding

ATSC 3.0 was designed to integrate the latest in audio/video encoding, while also supporting legacy encoding systems. The integration of HEVC H.265 video encoding into the standard introduces a significant leap forward in encoding technologies. HEVC already provides up to 50% gains over H.264, which is 50% more efficient than MPEG-2, the specified codec for ATSC 1.0.

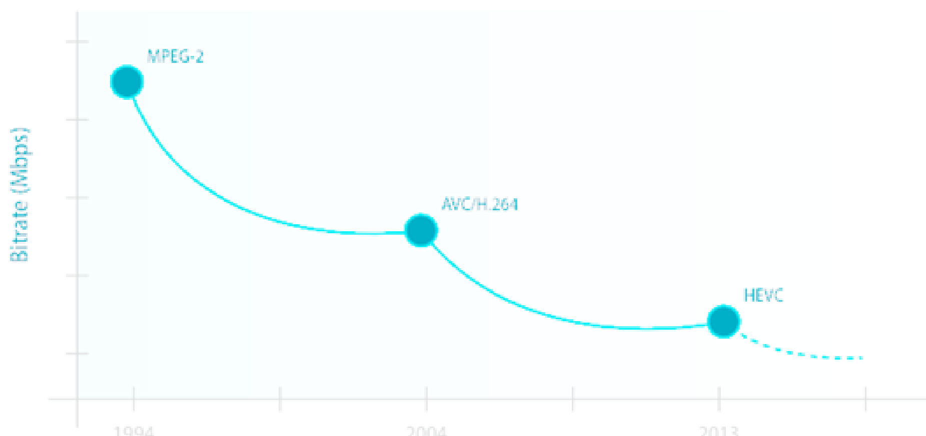

Figure 24: Efficiency gains of encoding systems

Source: Harmonic, Inc.

Today's IP-centric, high-density encoding platforms offer multiple benefits for ATSC 3.0 workflows, none more so than the ability to address the needs of multiplatform delivery on a single device. When combined with the ability to incorporate statistical multiplexing — a technology which balances video quality across channels by shifting the bitrate between those channels with low complexity to those with more — next-generation encoders bring significant benefits to broadcasters.

The move to software-based encoding gives broadcasters the flexibility to address multiple codecs and frame rates used in broadcast and OTT profiles all at the same time. From a single encoding unit, operators can support ATSC 1.0 with MPEG-2 HD today, and enable HEVC with broadcast and OTT profiles now and into the future. Moreover, such technology allows a faster turnaround to new software features, introducing additional gains in video quality improvements.

The A/V encoder takes in digital audio and video content from either SDI or SMPTE 2022-6 (uncompressed over IP) and produces encoded video frames at various resolutions. Depending on the licensed features and system capability, the content can be encoded in different formats and/or at different rates simultaneously. The encoded audio/video output is produced in the ISO/BMFF, which is “segment aligned” with the MPEG-DASH segment length. This alignment with MPEG-DASH allows a seamless transition capability between broadcast and broadband delivery of the multicast packets at the MPEG-DASH player/decoder. Ethernet interface is used to interconnect any number of encoders with the downstream MPEG-DASH packager.

Example: Harmonic Electra X2 (480i-1080p), Harmonic ViBE4000 (1080p-2160p), Ericsson MediaFirst Encoding Live, Ericsson MediaFirst Packager

MPEG-DASH Packager

ATSC 3.0 use of MPEG- DASH will allow broadcasters to use the same technology used by major OTT streaming services providers like Netflix, Hulu and YouTube. The MPEG-DASH (push) supports a variety of different MPD (the file manifest) and media formats. This design provides multiple ways to identify “chunk URLs,” while allowing different media containers and content encoding formats inside the same segment.

The DASH Packager wraps the incoming multicast encoded segments into the package output. In the case of ATSC 1.0, the frames are encapsulated into 188 byte MPEG-2 transport stream packets targeted specifically to the broadcast application. For ATSC 3.0, the output encapsulation is MPEG-DASH, which specifically targets the streaming market but also supports broadcast, which results in a set of files comprised of an MPD referencing multiple content segments. These files may simultaneously be forwarded to an origin server and Real Time (RT) ROUTE encoder, allowing the content to be delivered via broadcast transmission to first screen devices (including smart TVs and set-top boxes), and via the web to origin servers to support second-screen (smart phones/tablets and other IP-connected) devices. This forwarding typically occurs over IP-based File Transfer Protocol (FTP) or RTP.

The DASH packager is located between the output of the encoding system and the input to the ROUTE encoder. The inclusion of the DASH Packager is essential if DASH-ROUTE is used for content delivery. An alternative method supported by ATSC 3.0 is MMTP.

Example: Harmonic ProMedia Packager, KEEPIXO Packager

Signaling/Guide Manager and ROUTE Encoder

The signaling/guide manager and ROUTE encoder comprise a multi-function software application-based device. The signaling/guide manager is similar to the ATSC 1.0 Program System Information Protocol (PSIP) system and is configured via a web-based Graphical User Interface (GUI). This portion of the system receives various event listing information along with traffic and/or automation inputs to create ATSC 3.0 signaling and announcement data, which is then sent to the Gateway for integration into the broadcast.

Another function of this system is to manage advanced emergency alerting. This function is part of the Advanced Emergency Alerting (AEA) content manager which receives alert messaging from the station's EAS receiving and processing device. It also supports station-originated alerting. Associated with the AEA content manager is a content aggregator function that can pull in and manage the rich media content that supports advance emergency alerts.

The RT ROUTE encoder encapsulates the incoming DASH files into an ATSC 3.0 ROUTE stream and can be used to encode other near-real-time application content. The file data generated and/or processed by this system is then passed over to the ATSC 3.0 Gateway for final integration into the broadcast.

Example: Triveni Digital GuideBuilder® XM

Gateway, System Scheduler & Manager

In the writing of the ATSC 3.0 standards, the authors include the STL within the gateway, but in practical implementation the gateway and link are actually two separate systems. Several additions to the transmission architecture include a Signal Gateway and a System Scheduler and Manager, located at the station's master control. Like several other parts of the system, the gateway is a device that is similar to a multiplexer but has even more functionality. It aggregates all IP content signals and sorts them out by services into the various PLPs, and also integrates the signaling, announcement and control information from the system manager. It is the last device prior to the STL.

The gateway receives content files that have been ROUTE or MMP encoded. It then provides layer-two encapsulation for ATSC Link Protocol Base Band Format (ALPBBF) scheduling and framing. It receives multiple ROUTE (or MMTP) streams as IP, and produces baseband frames over a robust IP stream using UDP and RTP, plus Multicast RTP/UDP/IP, which is processed and output as STLTP. The Gateway also creates the bootstrap signal and establishes the modulation parameters that are handed off to the exciter.

System management at a base level is typically done with web based GUI, allowing the station engineers to program the number of PLPs, type of modulation of each PLP, amount of FEC applied to each, and selection of which program streams are assigned to each PLP, etc. Since these parameters must be coordinated with other up-stream system parts and configured so that the system emits a legal signal within the channel's bandwidth limitations, it is essential to get the configuration right.

As gateway products for ATSC 3.0 mature, they will incorporate a more sophisticated approach to system management that will allow external-automation-driven changes for the addition or deletion of services. This high level service manager, which is a software application, will also have standardized communications protocols to talk to other parts of the system such as the encoders, signaling and guide managers. This will keep all parts of the system coordinated as services change. Another part of this high level application will be a configuration manager, that includes a rules table and a calculating engine that can determine which requested system configurations are viable and legal. It will then provide "go" or "no go" feedback to the requestor.

Example: Enensys ATScheduler,TeamCast Stream4CAST

Studio to Transmitter Link

The STL requirements for ATSC 3.0 differ in several ways from those for ATSC 1.0. The MPEG transport stream in ATSC 1.0 is replaced by IP-based STLTP. Unlike the constant transport bit rate transmitted by ATSC 1.0, the ATSC 3.0 system allows a wide range of possible bit rates.

While the data rates needed for interconnection from the Gateway output to the Exciter input theoretically could require up to 157 Mbit/s of bandwidth (8MHz channel) to support any and all combinations of services and modulation modes, in practice, the likely maximum STL data rate for North American broadcasters using 6MHz channels would be limited to no more than 60 Mbit/s and in many cases the rate might not exceed 45 Mbit/s.

ATSC is standardizing around SMPTE 2022-1 Error Correction Coding (ECC) for the IP interface between the gateway and the exciter input. SMPTE 2022-1 ECC provides the necessary FEC to support high-quality STLTP over IP networks.

Stations typically employ one of three scenarios to link the station output to the transmitter input. In situations where the master control point and the transmitter are co-located, interconnect is rather simple, in that the link can be a simple CAT 5 or CAT 6 copper cable or an optical fiber link with simple short haul Modulator-Demodulators (MODEMS). In each case, the STL interconnect is a private circuit owned by the station.

Stations that have an STL with rented or leased fiber connectivity will require MODEMS that have the capability to operate over the correct fiber type provided by the third-party carrier. The MODEMS in either of the two fiber scenarios must support SMPTE 2022-1 as well as ATSC 3.0 STLTP.

Where stations are using microwave as the method for delivering the program stream transport to transmitter sites, there may be some significant changes from the current ATSC 1.0 STL solution. At a minimum, the microwave system will need to support DS-3 45MBit/s connectivity, and ideally the link should support 60 Mbit/s. Commercial microwave systems for IP transport are typically available in either DS-3 (45Mbit/s) or OC-3 (155Mbit/s) configurations from the same suppliers that provide equipment to carriers and network operators. As of this writing, no broadcast microwave manufacturers have released STL products that are specific to ATSC 3.0 requirements, but that is likely to change as ATSC 3.0 deployments gain traction.

In addition to the links(s) to the transmitter(s), many stations have direct links to Multichannel Video Providers (MVPD's). With ATSC 3.0, these links will be HEVC-encoded IP streams carrying the program(s) that are distributed by the MVPD. None of the signaling or STLTP information will be needed to accompany the program streams.

Example: Aviat Networks ODU600

Exciter

The ATSC 3.0 exciter/modulator takes in baseband IP frames in STLTP format delivered over the STL and emits a low level ATSC 3.0-compliant RF signal that is amplified by the transmitter's power amplifiers. The exciter also takes in RF samples from the transmitter output, and in some cases from downstream in the RF system. These samples are used to generate adaptive correction that is applied to the modulation process to improve the overall transmission performance.

A variety of exciter platforms have been developed over the past 20 years for ATSC transmissions. The newest generation of exciters has been developed using software-based modulation techniques and large Field-Programmable Gate Arrays (FPGAs), along with advanced adaptive pre-correction capability. These new exciters are extremely versatile and can be readily changed from one modulation to another very simply (usually via a software update only). They also have the capability to drive and optimize the performance for a variety of power amplifier types, including Inductive Output Tube (IOT) and solid state. Exciters are now available that can be set initially for operation on ATSC 1.0 and later converted to ATSC 3.0 without requiring any hardware changes.

Broadcasters currently using older model ATSC 1.0 exciters that cannot be upgraded to ATSC 3.0 may wish to consider replacing them with ATSC 3.0-capable exciters. In most cases, this can be accomplished without any issues. Items to consider when replacing exciters might include mechanical size and fit, electrical power requirements, cooling, RF output level, inputs, adaptive correction samples and levels, connector sizes, etc., and control/monitoring interfaces. In most cases, a suitable retrofit kit can be obtained from the exciter manufacturer that will provide all the needed items for a successful retrofit.

Example: GatesAir Maxiva XTE, Hitachi-Comark EXACT-ATSC

Transmitter

There are a number of factors to consider when evaluating transmitter needs for a transition to ATSC 3.0. In some cases the current transmitter may be suitable for ATSC 3.0, while in others, a new transmitter may be necessary or desirable.

Transmitters that are currently in ATSC 1.0 service vary in power level and design. Higher power UHF transmitters are a mix of IOT and solid state. All VHF ATSC transmitters in service today are solid state.

When evaluating the suitability of an existing transmitter for ATSC 3.0, there are a number of considerations:

- Exciter technology – Can the existing exciter be converted to ATSC 3.0, or is a new exciter needed?
- PAPR – There is a 2dB difference between the PAR of a transmitted ATSC 1.0 (8-VSB) signal and an ATSC 3.0 (OFDM) signal. This has several implications that should be considered.
- Maximizing Transmitter Power Output (TPO), versus replicating the coverage area.
- Adding Vpol to the antenna to improve mobile/portable reception. This directly impacts transmitter average power.

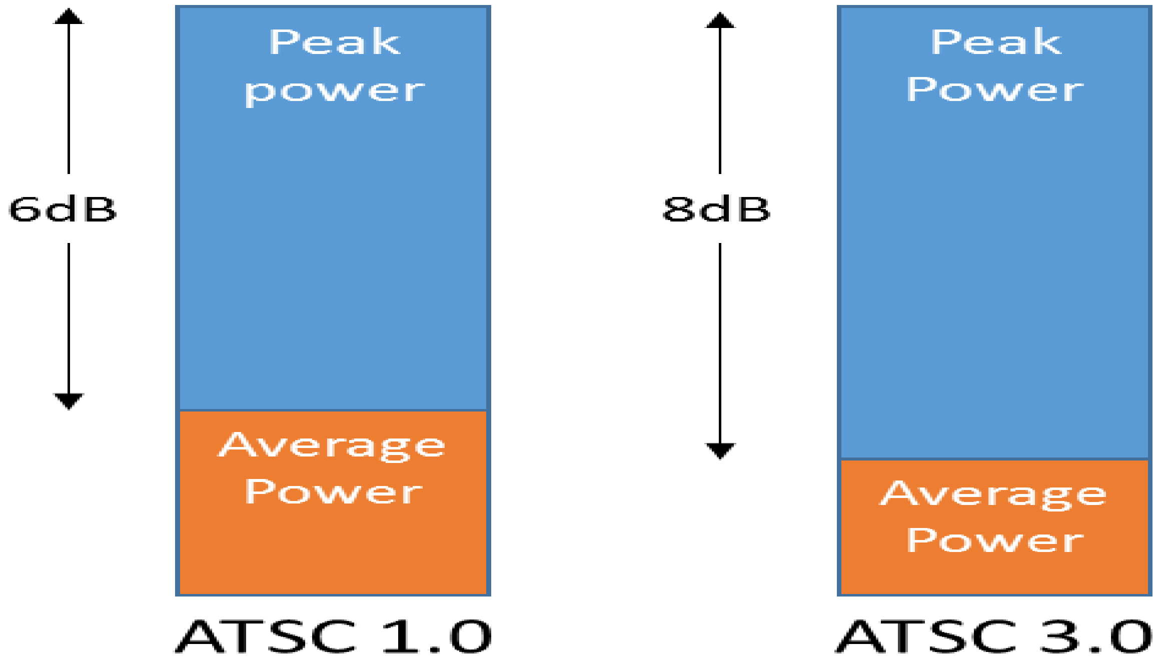

Figure 25: Peak Power Comparison ATSC 1.0 to ATSC 3.0

Source: GatesAir

The PAR difference between ATSC 1.0 and 3.0 is an important consideration. Most existing ATSC 1.0 transmitters are peak-power limited by virtue of their design and efficiency considerations; however the amplifiers are rated (and measured) in average RF output power. A typical ATSC 1.0 (8-VSB) RF envelope measured after the power amplifier and mask filter, meeting the FCC mask and SNR/EVM recommendations, will yield a 6dB PAR. An OFDM waveform, including ATSC 3.0, will yield an 8dB PAR. This 2dB difference impacts all items in the RF chain, including the transmitter. Depending on the RF amplifier design, attempting to increase the peak power may result in amplifier clipping that increases non-linear distortion to a point that is uncorrectable.

Therefore, a transmitter properly optimized for ATSC 1.0 will be operating at or close to its maximum peak and average power levels. If such a transmitter is modified for ATSC 3.0 modulation, the 8dB PAR for that waveform will limit the maximum average power, while maintaining the same peak power. This results in an average power that is 2dB lower than the original ATSC 1.0 average power.

While not all stations may require the same average power for ASTC 3.0 as they are currently operating with, many stations plan to maintain their current average power level after conversion to ATSC 3.0.

In some cases, it will be possible to increase the average power of a converted ATSC 1.0 transmitter back to its original average power level. To provide the extra 2dB peak power headroom needed, additional power amplifiers can be added, along with the appropriate RF system, electrical power and HVAC upgrades. Note that newer ATSC 3.0-ready solid-state DTV transmitters have been designed to operate with higher PARs and therefore can deliver the same average RF output power level for both ATSC 1.0 and 3.0. Broadcasters should consult with their transmitter manufacturers to verify this capability.

Of course, other factors will also determine the required ATSC 3.0 power level. In some cases, not maximizing to the ATSC 1.0 Effective Radiated Power (ERP) and Transmitter Power Output (TPO) may be acceptable. Since the ATSC 3.0 standard allows for a wide variety of modulations (i.e., QPSK, 16QAM, up to 4096QAM) and many FEC rates, etc., there are almost infinite possibilities. Much will depend upon the desired payload, Carrier-to-Noise (C/N) and reception type (i.e., fixed rooftop, indoor, portable, and mobile). At this early stage it may not be clear what parameters are required and what average power level will be needed. In general it is best to allow for maximizing the average power.

Another item to consider when figuring transmitter size for ATSC 3.0 is Vpol to the antenna. This will help provide better reception for portable/mobile devices, as well as indoor devices that may have less than ideal or moving antenna orientation. Although there are different opinions as to how much Vpol is optimal, a range of 25% to 33% is recommended for planning purposes. This will increase transmitter power by the same percentage. Please note that ERP is based only on the Horizontal Polarization (Hpol) power; therefore a higher power transmitter may be required (2x in the case of 100% Circular Pol) should a vertical component be added for ATSC 3.0.

Example: Gates Air Maxiva, Hitachi-Comark Parallax

RF System: Filters, Combiners and Transmission Line

The ATSC 3.0 physical layer is based on OFDM principles. This results in two areas of interest when applied to RF systems. The occupied bandwidth and PAPR are both greater than those occurring in 8-VSB systems. The occupied bandwidth impacts the mask filter while the PAPR needs to be considered in all components in the RF system, transmission line and antenna.

Initial tests conducted by Meintel, Sgrignoli and Wallace (MSW) have shown that ATSC 3.0 into ATSC 1.0 co-channel and first adjacent channel interference are very similar to those for ATSC 1.0 into ATSC 1.0, as long as the FCC emission mask requirements prescribed for ATSC 1.0 continue to be met. A condition for ATSC 3.0 acceptance by the FCC is that interference planning factors do not need to change. Thus, meeting the existing emission mask is imperative. The occupied bandwidth of the ATSC 3.0 signal can be as broad as 5.83 MHz (97%) in some configurations, significantly wider than the 5.38 MHz (90%) of ATSC 1.0. How will this broader signal be affected by the RF system?

Most mask filters in use for single-channel ATSC 1.0 operation are six-pole filter designs with a sufficiently broad amplitude response to pass the slightly wider ATSC 3.0 signal with no amplitude response distortion. Shoulder performance will, as in ATSC 1.0 operation, be determined primarily by the transmitter's non-linear correction. Eight-pole filters should be swept before applying the ATSC 3.0 signal to make sure they are correctly tuned, and to determine if one of the five carrier reduction coefficients described below should be applied.

There are two advantages that ATSC 3.0 offers in relation to occupied bandwidth. First, OFDM signals are largely immune to the typical linear distortions seen in the mask filters in RF systems. Each carrier in the signal occupies such a narrow bandwidth that group delay across the carrier is irrelevant, as is group delay across the channel. Second, there is a feature available in ATSC 3.0 that allows an operator to select a carrier reduction coefficient. The number of carriers in a given FFT can be reduced in five defined increments, thus reducing the occupied bandwidth if required by certain filter implementations; for example, in adjacent channel combiners. Using carrier reduction will, of course, reduce the channel capacity by a modest amount.

Adjacent channel combiners incorporating a mask filter typically use eight-pole filter designs to provide the required attenuation at the band edge that is adjacent to both channels. These filters have a roll-off characteristic much steeper than six-pole designs, and there may be some attenuation of the ATSC 3.0 band edges as a result. If required, the carrier reduction technique can be applied to limit the roll-off of the edges. It is unlikely that such roll-off will significantly affect the ATSC 3.0 signal fed into the narrow band input of the combiner. The carrier reduction technique may be more useful in reducing adjacent channel spillover further than the filter alone.

Since it is likely that the FCC will permit ATSC 3.0 operation at the same average ERP as that for ATSC 1.0, the consequence of the increased PAPR of the ATSC 3.0 signal is that peak power levels in the ATSC 3.0 RF system will be as much as 2dB higher than those in ATSC 1.0 systems if the transmitter can deliver those peaks. If not, it may be necessary to reduce the average power of the transmitter until it can pass the higher peaks without significant distortion. However, because of the conservative safety factors applied to filter and transmission line designs, it is not expected that PAPRs of 10 or 11dB will be a problem at typical average operating power levels. In all cases, broadcasters should verify with their RF equipment suppliers to confirm the ATSC 3.0 ratings of installed ATSC 1.0 equipment or planned equipment being considered for conversion.

Three factors might necessitate a change of a station’s existing transmission line.

- If a station is repacked to a new channel and the existing line has too high of a VSWR as a result of the line section lengths, a replacement will be necessary. Stations that are likely candidates for repack should sweep their transmission lines to determine which channels could operate on existing lines.

- If the station’s existing transmission line is operating near the rated power capacity for ATSC 1.0, the additional 2dB peak power needed to maintain equivalent ATSC 3.0 average power might require transmission line replacement.

- The addition of Vpol by changing out the station’s antenna system will require additional RF power to be delivered to the antenna. This could be via a single transmission line or the addition of a second transmission line as determined by the antenna design. If the solution requires using a single transmission line, the power increase through the line could be up to an additional 3dB beyond ATSC 3.0 Hpol-only operation, depending on the Vpol-to-Hpol power ratio. If a second transmission line is required for Vpol, it is likely that this line will be identical in all aspects to the line used for Hpol.

In any of these cases, there will be an impact on the tower loading, and a study and review will be needed to verify that the transmission line changes or additions can be accommodated.

Example: Dielectric EIA Line, DigiTLine™, EHTLine™, fixed and tunable High Power mask filters

Antenna

The critical factor to evaluate when considering ATSC 3.0 antenna operation is the signal’s peak power. It is likely to be as much as 2dB higher than the peak power of an ATSC 1.0 signal operating at the same average power. Antenna manufacturers can provide confirmation of the antenna's peak and average power ratings. Therefore, the manufacturer should be consulted early in the conversion planning of an existing 1.0 antenna. Likewise, if a new antenna is considered as a consequence of the repack, ATSC 3.0 ratings should be considered in the selection process.

The increased capability and flexibility of ATSC 3.0 creates the need to consider additional items before deciding on a new antenna. For example, ATSC 3.0 has operating modes that are ideally suited to signal reception in portable, indoor devices and moving vehicles such as cars or trains. In both cases, a small antenna will be used, and it has been shown that reception is enhanced by the addition of a vertical component (Vpol) to the horizontal component (Hpol). The FCC licenses TV stations on the horizontal ERP only but allows up to an equal amount of power to be radiated in the Vpol.

Extensive field testing has shown that the actual amount of vertical component is not critical. Vpol power of between 30% and 100% of Hpol power improves the fade margin to reception by a small antenna of varying orientation by as much as 8dB in heavy scatter environments. Vpol equal to Hpol is considered full Circular Polarization (CP). Regardless of the actual amount, additional transmitter output power is required to feed the vertical component. The cost to provide the additional output power is not trivial. In addition to potential capital procurement, more output power directly results in greater operating expense in the form of utility bills.

If a station is not required to move to a new channel during the repack program and continues to operate on its current channel, adding Vpol is an expensive option since a new antenna, and potentially a new transmitter, will be required to take full advantage of the ATSC 3.0 capabilities. Note that adding Vpol is not mandatory for ATSC 3.0 operation. Non-repacked stations can still transition to ATSC 3.0 with horizontally polarized antennas and realize many of the benefits. If a station is required to change channel during the repack process, consideration of adding Vpol should be factored into the repack planning. As noted above, this consideration will have to take into account the actual antenna design and the output rating of the transmitter.

Another, perhaps not obvious, consideration for a new antenna is the capability of field conversion for increased null fill at a later date. ATSC 3.0 is, in certain operating configurations, capable of supporting data-intensive services that typically require higher C/N ratios. One proposed business model for ATSC 3.0 is to provide such high capacity, high C/N services in areas close targeted to the most densely populated areas of a market by adding null fill, then making up for the signal reduction at the periphery with the addition of transmitters in a SFN. Such field conversions can readily be accomplished, but the antenna must be designed specifically with this future service in mind. Stations should note that the FCC's repack reimbursement rules only allow funds for like-for-like expenses. Thus, adding V-pol or the field conversion capability may not be covered by the repack reimbursement program, but stations should consider supplementing the reimbursement amount to make the purchase at the time of repack, to avoid much greater expense in the future.

In addition, some stations will require auxiliary antennas. Stations considering auxiliary antennas as repack facilitators should also consider that auxiliary antennas may also be used for ATSC 3.0 services as back-up to the main antenna. Consideration should also be given to the addition of V-pol for the same reason as it is beneficial in the main antenna – increased reception margin.

In all cases, stations should contact their provider to confirm the peak and average power ratings of existing or planned equipment.

Example: Dielectric FutureFill™ Slot and Panel antennas, WB Auxiliary Antennas

Implementation Models to Optimize Coverage

One of the primary reasons identified by broadcasters for the development of ATSC 3.0 is the need to optimize OTA transmission coverage to reach mobile and portable receiving devices, as well as home TV sets with indoor antennas. Achieving successful reception on devices with less than optimum receiving antennas requires the deployment of multiple solutions that will improve the link performance and availability. Essentially there are three solutions that are available to broadcasters to optimize OTA coverage. The first is to select modes of modulation and coding that lower the required signal threshold (C/N) for successful reception. The second is to add Vpol radiation to the main transmission system, and the third is to deploy distributed or supplemental transmission points within the station's assigned coverage area. A combination of all three will prove to be the best approach to improving and optimizing a station's coverage within its assigned market.

Vertical or Circular Polarization

For stations that do not currently have Vpol antennas, the addition of vertical or circular polarization (see Antenna section on page 53) represents a major step in signal availability improvement. If repack is in the station's future, much of the expense of a new antenna, and the installation costs for this change, can be covered in the repack reimbursement as opposed to the station absorbing the full cost to make this change at a later time.

There are some related factors that must be considered before making this change. It is likely that antennas with Vpol will have some increase in weight and wind load over that of an Hpol-only antenna. Also, depending on the design of the antenna, the transmission line requirements might either increase the size of a single transmission line or require a second transmission line to deliver the RF power to the Vpol section of the antenna. As previously mentioned, Vpol will require additional transmitter power that could range from a 30% up to 100% increase in output power, depending on the ratio of Vpol to Hpol that is selected.

Future Deployment of SFN Configuration

It is widely held that ATSC 3.0 SFNs will enable new opportunities for U.S. broadcasters in the form of mobile reception and "hyper local" content insertion.

Since ATSC 3.0 is an OFDM-based system, it will also allow the use of SFNs when inserting an appropriately-sized guard interval. There are many options, allowing, in principle, unlimited SFN size using a guard interval up to 600 us (6 MHz). The capability to provide a service even at negative C/N might also mean that the guard interval length can be shortened even in very large SFNs. That standard's performance in a SFN configuration remains to be fully tested, and currently such SFN trials are being conducted by broadcasters, including Sinclair in the Washington, D.C./ Baltimore area. This project was intended to be a launching point for ATSC 3.0 SFN implementations in the U.S. using commercially available network planning software, and evaluating reception results in the service area. (See Addendum A for more detail).

SFNs are common in OFDM-based broadcast systems throughout the world. Each symbol of the OFDM signal is preceded by a copy of the end of that particular symbol, known as the guard interval. If an echo of the original signal arrives at the receiver with a delay less than the guard interval, all signal components within the receiver integration period come from the same symbol and do not generate inter-carrier or inter-symbol interference. Multiple transmitters emitting identical signals at identical frequencies will appear as echoes to a receiver. If the transmitter location is selected such that any appreciable signal arrives at any receiver within the guard interval, the reception will be enhanced throughout the service area. U.S. broadcasters have not been able to fully enjoy SFN benefits because of the single carrier 8-VSB system's limitations, especially the lack of a guard interval.

There are essentially three approaches to deploying SFN transmission systems. The first, similar to wireless carriers' and European broadcasters' networks, uses a network of transmitter sites across an area, with each site at similar power levels and antenna heights, and with both typically less than the current high power tall tower model. Depending on the market size, the array of transmission sites might range from 6 to more than 12 to cover a market.

A second approach is to add SFN booster sites to supplement the existing high power tall-tower transmission system. These sites are best positioned at the outer edge of a station's coverage area, and employ directional antennas that beam the signal towards the center of the coverage area without expanding the authorized coverage. This method will raise the average signal level and availability across the station's coverage area, and provide diversity transmission that will enhance indoor reception. Typically 3 or 4 of these sites would be needed to fill out the entire coverage area. The most cost effective way to implement this approach is for all (or most stations) in one market to share these booster sites with a common tower, broadband antenna, transmission line, combiner and support infrastructure such as the transmitter building. Each station would have its own transmitter feeding into the combiner. These sites could be owned and managed by a vertical real estate company to reduce individual station's capital investment.

The third deployment model is applicable when station owners have stations in adjacent markets or are able to extend their stations' coverage beyond the existing market by operating two or more high power transmission sites on the same frequency. This is similar to the test model that Sinclair deployed between Baltimore and Washington D.C.

Although SFN planning is not as impactful to a broadcaster's main transmit site when planning for repack, SFN planning should be considered at the same time. If there is an opportunity for licensees to unite and share a broadband antenna for their main transmit facilities during repack, this will improve the possibility of licensees in each market to share SFN facilities.

SFN implementation and build out considerations:

- Use of standardized SFN RF design software to achieve desired market coverage and business objectives;
    - *Example: Acrodyne Services PROGIRA® plan;*
- Evaluate existing wireless and smaller broadcast-compatible towers for SFN use to avoid new tower construction;
- Analyze small tower/low power vs. taller tower/medium power SFN approaches to adjust for existing tower use and optimize coverage service characteristics;
- Consider shared SFN infrastructure to minimize CAPEX and OPEX costs.

Transitioning Stations in a Market to ATSC 3.0

Unlike the transition from analog to digital that the industry experienced between 1996 and 2009, the transition from ATSC 1.0 to ATSC 3.0 will not be supported by the use of a second channel for each station. A solution to this problem lies in a fundamental aspect of the spectrum repack, with the option for stations to enter a channel sharing agreement (CSA) and share the same spectrum. This option, combined with a future transition to ATSC 3.0, can lead to a number of interesting scenarios that will benefit all participants.

A group of industry leaders and experts have put together a concept of temporary channel sharing that supports the creation of an initial ATSC 3.0 station known as the “Lighthouse Station” that will seed the market with 3.0 services coming from multiple stations in that market. This of course requires a business collaborative between partner stations involved in creating this “Lighthouse Station.” An ideal theoretical model would be collaboration among the five primary network stations in a single market. The Lighthouse Station will have its ATSC 1.0 signal carried by the other partners in the business collaborative.

It should be noted that the current ATSC 1.0 PSIP system can support this channel sharing by preserving the stations’ branding and recognition of the virtual channel. It will, however, require that consumers rescan their receiving devices when a station shares its ATSC 1.0 stream for the first time.

Over time, as ATSC 3.0 consumer receiving devices are widely deployed, additional stations within the lighthouse collaborative will convert their transmissions to ATSC 3.0, while the remaining stations carry some of the services from the converted stations in ATSC1.0. As more stations are converted to ATSC 3.0, the available payload capacity will allow the converted stations to carry many of the services outlined in this Guide.

When consumer adoption of ATSC 3.0 is sufficiently high, all stations in the market may want to convert to ATSC 3.0. One possible option is for the last station in the market to remain on ATSC 1.0 for a period of time while carrying services from each of the converted stations. This concept of a “Night Light Station” would provide some ATSC 1.0 service, thus not stranding any consumers who don’t have ATSC 3.0 receivers.

There are several points of technical consideration for stations planning to enter into such collaborative business arrangements. For those stations contributing program streams to the Lighthouse Station, they will need an HEVC encoder with IP output for each of the contributed streams, and a reliable IP link of sufficient bit capacity to the hosting station.

It is likely that each contributing station would begin its 3.0 stream by operating in an advanced video format such as 1080p60 and perhaps even with HDR. The limitation on the video format would be driven by the number of individual streams that the Lighthouse would carry and the overall parameters of the emitted 3.0 signal. This was previously discussed in the section on Understanding Payload Trade-offs on page 44.

For those stations remaining on ATSC 1.0 but carrying additional ATSC 1.0 streams from the Lighthouse or other stations that have converted to ATSC 3.0, they will likely need to engage in a distributed architecture to create a geographically dispersed statmux pool, where each station hosts their own encoders and contributes to a shared transport stream that is physically transmitted by the 1.0 host station.

The available technology is able to support the transition when stations engage in these types of collaborative sharing arrangements. Transition success will ultimately come down to creating a viable temporary business arrangement among the partners.

Config. No.	Opportunity	Multiplex Capacity	Targeted Receivers	Service Assignments	Channel Loading (BCN On)	PLP Capacity (Mbps)	AWGN SNR (dB)	Comments
1	Multicast HDR/WCG from Single Stick TDM	6 (see note)	Fixed	6 HD+ Multicast Video	PLP #1-6 : 32K FFT; 148 usec GI, 256QAM, 64800 LDPC, 9/15 FEC, 250 msec Frame	~ 4.0	15.50	HD+ w/HDR & WCG Lighthouse Operation
	Audio Services		(similar to today)	Audio/Caption	PLP #7-12: 32K FFT; 148 usec GI, 64QAM, 64800 LDPC, 10/15 FEC, 100 msec Frame	~ 0.3	12.88	
		Stations w/ Individual PLP's		Signalling	PLP #13: 32K FFT; 148 usec GI, 64QAM, 64800 LDPC, 2/15 FEC, 100 msec Frame	0.128	-0.26	
2	Multicast HDR/WCG from Single Stick TDM	6 (See Note)	Fixed	6 HD+ Multicast Video	PLP1: 32K FFT; 148 usec GI, 64QAM, 64800 LDPC, 11/15 FEC, 250 msec Frame	21.00	14.28	HD+ w/HDR & WCG Lighthouse Operation
	Audio Services		(similar to today)	Audio/Caption	PLP2: 32K FFT; 148 usec GI, 64QAM, 64800 LDPC, 10/15 FEC, 100 msec Frame	2.40	12.88	
		Stations w/ Shared PLP's		Signalling	PLP3: 32K FFT; 148 usec GI, 64QAM, 64800 LDPC, 2/15 FEC, 100 msec Frame	0.128	-0.26	

Note: Lighthouse Transmitter Configuration for Transition - Multicast 6 stations on single RF Channel [Config 1 is Estimated since 13 PLP Calculator is not available].

Figure 26: Lighthouse operation configurations

Source: Meintel, Sgrignoli & Wallace, LLC

Planning for ATSC 3.0 as Part of Spectrum Repack

With the television spectrum auction currently underway, many stations will be facing a change of channel assignment known as repack. Recent estimates indicate that the number of stations facing repack could range from 800 to more than 1,000. A large percentage of these stations will require replacement of some or all transmission components, starting at the exciter and progressing through to the antenna. Those stations facing transmission line and antenna changes will also likely encounter some level of impact on the supporting tower structure.

Transition timetables will be unknown until the auction closes. Under the original FCC schedule, stations were to be assigned their new channel around the end of October 2016. That date now looks like it will slide into 2017, as the spectrum auction is likely to go to at least multiple rounds before it closes.

In April 2016, four industry organizations, NAB, Consumer Technology Association (CTA), The Association of Public Television Stations (APTS) and the AWARN Alliance filed a request to the FCC to adopt and authorize stations to broadcast using ATSC 3.0 technology. Shortly after, the FCC issued a Public Notice (PN) asking for comments on this request. The broadcast industry is requesting for a Notice of Proposed Rule Making (NPRM) to be issued in late 2016 or early 2017.

All stations required to move to a new channel assignment are eligible for repack reimbursement, with the FCC's fund paying for equipment that is equivalent to their current equipment, plus approved costs for planning, and installation and removal of equipment. With the industry conversion to ATSC 3.0 following shortly after the repack, it makes good economic sense to plan for the ATSC 3.0 conversion at the same time as planning to move to a new channel assignment. Proper planning could save the cost of doing tower work twice.

As previously stated, ATSC 3.0 transmission equipment components (transmitter, RF system and antenna) will require the ability to handle a 2dB higher PAR than the current ATSC 1.0

system for an equal level of average RF power. While the repack compensation fund will only pay for current equivalent components, it makes long-term financial sense for stations to fund the cost difference when repacking, to pay for the additional peak power capacity and for improvements such as the addition of Vpol to their antennas. At the very least, a transmitter should be selected that is easily field-upgradable later to the required power level.

Additionally, the cost of the installation services in most situations would be the same for installing the upgraded components as it would be to install components that are only rated for ATSC 1.0 power levels.

The long-term savings gained by making the upfront investment includes a significant portion of the cost of the transmission equipment and virtually all of the engineering and installation services.

Up to this point, this Guide has focused on the equipment and services aspects of repack combined with ATSC 3.0 planning. There are also considerations that need to be made for the tower structure, site and operational impact that would support the decision to plan for ATSC 3.0 as part of the repack process.

Preparing the Tower and Tower Site

The incentive auction and subsequent spectrum repack creates a highly complex set of circumstances for broadcasters and the businesses that support the television industry. Among these circumstances are the needs to mitigate risk associated with tower work, and minimize the amount of service disruption that will occur when moving service to a new channel assignment.

Any change to a licensee's transmission system mounted on a tower, or on the ground, that requires larger or higher power RF equipment may require modifications and changes to the tower and ground-based support infrastructure. Broadcasters that are not moving to new channel assignments may also want to consider ATSC 3.0 future-proofing, whether they are part of a current broadband antenna system with stations being moved to a new channel, or join a new broadband antenna project with enhanced ATSC 3.0 capabilities.

Whether a broadcaster owns its tower or leases tower space, the following should be investigated as part of ATSC 3.0 planning during repack:

- Evaluate tower structural loading and physical space to consider the impact of any change in transmission equipment on the tower:
    - Increase in weight and wind load for new single frequency or broadband with added Vpol component; and
    - Increase in transmission line size or multiple lines to handle additional power for vertical and/or variable polarization antennas.
- Evaluate building and mechanical system for transmitter power increase due to addition of Vpol component:
    - Evaluate floor space to add larger transmitters while staying on the air;
    - Evaluate site mechanicals to support larger transmitters and filters;
        - Transmitter cooling
        - HVAC

- Electrical power
- Generator

By evaluating and modifying repack construction plans to include the changes required to support ATSC 3.0 RF transmission equipment, broadcasters should keep the following in mind when planning for ATSC 3.0 and repack:

- FCC will reimburse repack licensees for eligible new RF system expenses;
- FCC will reimburse repack licensees for eligible tower and transmission site modification expenses;
- Eliminate changing RF equipment twice;
- Eliminate modifying tower and transmission site infrastructure twice;
- Minimize OTA downtime by consolidating tower repack and ATSC 3.0 on-tower work;
- Minimize risk by decreasing the time crews and construction equipment are on the tower.

For those broadcasters remaining after the reverse auction, proper planning during the repack period creates a way to develop, construct and operate transmission facilities that will enhance the new technology attributes of ATSC 3.0 and hence "future-proof" their commercial success.

Conclusions

Because of ATSC 3.0's flexibility and status as a hybrid broadcast/broadband platform, it is difficult to describe all of its possible applications, network configurations, and business opportunities in one place. This is not a commentary on the incompleteness of this Guide, but is a testament to the platform's promise as a tool for industry evolution. There are so many possible combinations of services and network configurations that it is not reasonable to describe them all in this primer.

As finalization of the standard draws closer and imaginative broadcasters test its abilities and the business opportunities they spawn, there will be much more to describe and document. Many of the standard's features and tools have yet to be tested, which is due not only to the platform's infancy but also to the importance of giving sufficient study to all it can enable.

It may be in its infancy but the sense of urgency for evaluating and planning for its implementation is real, as broadcasters navigate a much-changed media landscape, as well as a near-term industrywide repacking of the 600 MHz band. Because broadcasters can build future proofing into their potential channel changes, 3.0 infrastructure will begin soon.

General Timeline

The platform's testing has already begun, although the complete ATSC 3.0 suite of standards won't be finalized until the fall of 2016. Live tests in Cleveland, Madison, Baltimore/Washington D.C., and Raleigh have tested parts of the system (details in Appendix A) such as terrestrial broadcasts to stationary devices and moving vehicles, as well as the advanced Emergency Alert

System and advanced coding. In Baltimore/Washington D.C., Sinclair Broadcast has tested a basic SFN network (details in Appendix A).

Planning for services from a business perspective, petitioning the FCC for regulatory permission (on a voluntary basis) to transmit ATSC 3.0 signals, and creating educational programs to inform regulators and policy makers of the standard's benefits have all begun in earnest. Pioneering broadcasters such as Sinclair Broadcast, Capitol Broadcasting, and Tribune Broadcast have worked with equipment suppliers to "turn on" the first transmissions. This has resulted in prototypes, and in some cases, commercially available equipment designed to build ATSC 3.0 networks. Meanwhile, by the end of 2016 or early 2017 (depending upon when a new 600 MHz band plan is generated) broadcasters will be placing orders for equipment needed for new channel assignments. Therefore, orders for ATSC 3.0-ready equipment are imminent.

In the meantime, South Korean broadcasters have officially adopted ATSC 3.0 for its next-generation broadcast/broadband platform, and plan first commercial broadcasts for the spring of 2017. LG Electronics will likely be the first to supply the first consumer receivers for the South Korean transmissions that will highlight the 2018 Winter Olympics.

Broadcast equipment manufacturing will likely begin in 2017, as continued testing and the South Korean activity primes the R&D and manufacturing processes necessary to begin production. In addition, many elements, such as coding, IP encapsulation, and system interfaces rely on open standards, which will shorten the time-to-market for equipment.

Once the FCC proceeds with the NPRM process and receives input from respondents, official rule making will take place to allow stations to adopt ATSC 3.0 as their mode of digital broadcasting. The first commercial U.S. broadcasts will likely begin in 2018, with more adoption once broadcasters are completing transitions to new channels during the Repack Program. It is important to note that until the Spectrum Auction has closed, it will be difficult to identify timetables, until the FCC creates the repack plan in accordance with the number of stations voluntarily vacating their spectrum.

ADDENDUM A: REVIEW OF ATSC 3.0 FIELD TESTING

Introduction

Each field test case study was written by direct participants within each test. The case studies encompass three locations – Madison, Cleveland, and the Baltimore/Washington D.C. area, and they present varying levels of detail in accordance with available test results provided by the authors.

In addition, Capitol Broadcasting, which operates WRAL-TV in Raleigh-Durham, N.C. became the first commercially-licensed television station to broadcast its news using the ATSC 3.0 platform in June, 2016. The NBC affiliate transmitted the ATSC 3.0 signal by airing the WRAL News at Noon on channel 39.1 while simultaneously airing a documentary shot, edited and post-produced in 4K/UHD HDR on 39.2.

Madison

As the ATSC 3.0 physical layer evolved based on decisions made within the S32 specialist group, numerous simulations were demonstrated to show how the physical layer would perform. While simulations are a good start toward understanding system performance, going beyond simulations and conducting actual field testing was critical, as it would subject the selected waveforms to actual conditions experienced when transmitted within the television broadcast spectrum.

Figure 27: Routes for field testing in Madison, WI

LG Electronics, working with its Zenith Labs subsidiary and partner GatesAir, constructed an early modulator platform and receiver based on FPGA technology that would support transmission testing. The initial goal was to compare the proposed physical layer against the existing ATSC 1.0 A/53 and A/153 performance. In October 2014, Quincy News Inc. volunteered its station in Madison Wisconsin (WKOW) for use during a three-hour overnight period that coincided with the annual Wisconsin Broadcast Workshop. WKOW operates on channel 27 with 800kW ERP.

The Madison area offers a variety of terrain and reception situations ideal for testing. Indoor reception was tested at the University of Wisconsin's Kohl Center and at a local video production house located within an all-metal structure. Mobile and outdoor testing was conducted along several routes including downtown Madison, a route running 53 miles southwest of Madison, and a route running more than 40 miles northwest of Madison.

LG used its field measurement vehicle and a portable test rig to measure signal levels, demodulate the transmitted signal and to capture the RF transmissions at and along the various test routes. Prior to the test of the physical layer, the same locations and routes were measured and captured while the station was transmitting both ATSC A/53 and A/153 signals. During the Madison test of the physical layer, there were three digital pipes utilized. DP0 had an equivalent 15 dB C/N to ATSC A/53 but with 36% additional bit capacity. DP1 was configured to be equivalent to A/153 at ¼ rate but with 2.5X bit capacity, and DP2 was configured to perform at about -1.5dB C/N. The signals tested contained some of the proposed error correction codes, but did not have a bootstrap or some of the more advanced coding that had not yet been decided by the ATSC S-32 specialist group.

More than 16,500 data points were collected during the evening testing, with some remarkable indoor reception points within downtown Madison buildings. The results were encouraging. However, a data collection issue was discovered, which likely skewed the collected information.

Cleveland

Having limited access to the station in Madison only during selected overnight hours was a handicap that prevented the ability to fine-tune and rerun test points to make more detailed signal performance comparisons.

A more suitable test site was found in Cleveland, Ohio that would allow continuous operation while testing, without the need to support the commercial station's activities. The transmitter, transmission line and antenna were still in place but had been dark since 2009 when WJW returned to its original Channel 8 assignment at the completion of analog broadcasting. An agreement was made among Zenith Labs, GatesAir and Tribune to gain access for use of the facility. After some negotiations with an adjacent channel LPTV operator, a Special Temporary Authorization (STA) was granted by the FCC to operate the facility using a preliminary version of the proposed ATSC 3.0 waveforms. Like Madison, no bootstrap, LDM or advanced coding was implemented on the test platform.

During May and July 2015, testing was conducted from a Cleveland broadcast site using a similar system configuration as in the Madison tests. This test was executed by the same team that conducted the Madison test (see Madison section above).

The data capture issues that occurred in the Madison test were resolved prior to the Cleveland tests. In addition, the station operated with A/53 and A/153 emissions to serve as a comparative benchmark.

Multiple mobile routes were characterized and measured, including one 50 miles to the southwest, another 40 miles to the east and one 25 miles to the south of the Cleveland transmitter site, which is located in the suburb of Parma. Fringe reception was also captured on a path running east to west at about 25 miles south of the transmitter, and there were downtown mobile and indoor fixed capture points selected. Overall, there were more than 18,000 data collection points captured for each of the modes of operation, including both of the ATSC 1.0 modes.

Figure 28 illustrates the routes utilized in the Cleveland tests. The testing shows that the ATSC 3.0 physical layer operating in the DP1 mode was very comparable to the A/153 M/H mobile mode in ATSC 1.0, but the DP2 mode indicated a significant performance improvement over anything that ATSC 1.0 could deliver in either mobile and fixed reception.

Figure 28: Routes for field testing in Cleveland, OH

Figure 29 shows the comparative reception performance for all three preliminary ATSC physical layer modes tested against ATSC A/153.

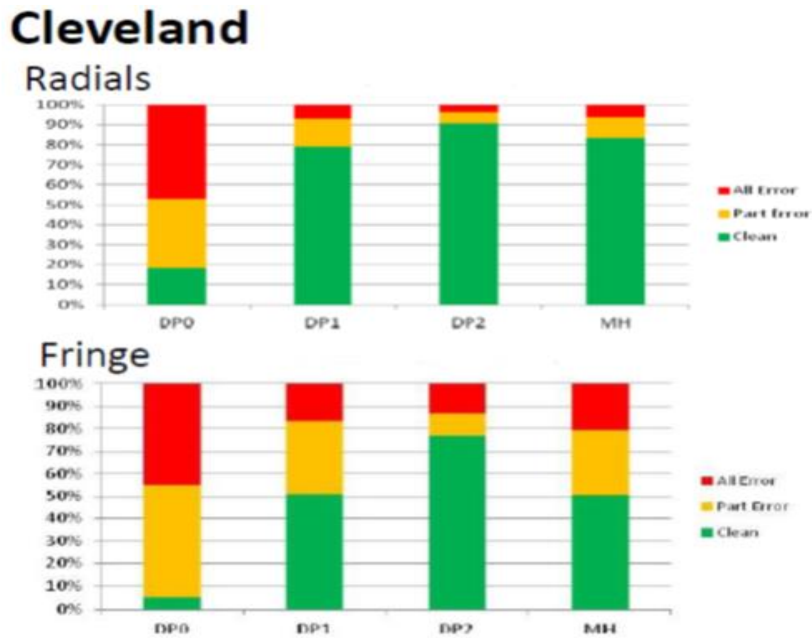

Figure 29: Reception performance test results in Cleveland, OH

Real-world testing of the physical layer has proven to be a successful confirmation of the projected performance that came from the various simulations and calculations that were driving decisions within the S-32 specialist group as it created the ATSC 3.0 physical layer.

The Cleveland facility operated under an STA granted by the FCC during the May and July 2015 tests. Looking toward future testing of the ATSC 3.0 system in Candidate Standard or Proposed Standard form and the need for more in-depth types of tests, GatesAir and Tribune Broadcasting applied for an experimental license, which was granted in October of 2015 with the call sign W19XJY. Testing was performed in July of 2016, with early results being internally analyzed as of this writing. The test used the A/321 and A/322 current versions that are being balloted within ATSC. Once LG, Zenith Labs and GatesAir complete their testing, the Cleveland site and experimental license will be transferred over to the NAB, and opened up for additional testing of both transmission and receiving devices designed for ATSC 3.0.

Baltimore/Washington ATSC 3.0 SFN Field Tests

The Baltimore/Washington ATSC 3.0 project was conceived with several objectives in mind:

- To provide a test bed for evaluating prototype next-generation broadcast technologies;
- To observe the performance of equipment initially designed for ATSC 1.0 service with ATSC 3.0 signals;
- To build a simple two station SFN to evaluate ATSC 3.0 planning software and start the process of defining SFN system design issues; and
- To expand the SFN and, if appropriate, evaluate on-channel repeaters.

SFNs are common in OFDM-based broadcast systems throughout the world. Each symbol of the OFDM signal is preceded by a copy of the end of that particular symbol, known as the guard interval. If an echo of the original signal arrives at the receiver with a delay less than the guard interval, all signal components within the receiver integration period come from the same symbol and do not generate inter-carrier or inter-symbol interference. Multiple transmitters emitting identical signals at identical frequencies will appear as echoes to a receiver, and if transmitter location is selected such that any appreciable signal arrives at any receiver within the guard interval, reception will be enhanced throughout the service area. U.S. broadcasters have not been able to enjoy the benefits of SFNs fully because of the limitations of the single carrier 8-VSB system, and in particular the lack of a guard interval. Although there have been several attempts to build 8-VSB SFNs, the limitations imposed by poor multipath receiver performance have limited their success and application.

It is widely held that ATSC 3.0 SFNs will enable new opportunities for U.S. broadcasters in the form of mobile reception and "hyper local" content insertion. This project was intended to be a launching point for ATSC 3.0 SFN implementations in the U.S. using commercially available network planning software, ProgiraPlan, and evaluating reception results in the service area.

The experimental STA was granted for operation on Channel 43 on August 24, 2015, at two sites: the TV Hill tower currently in use by both WBFF and WNUV in Baltimore, and an unused tower at the WRC site in northwest Washington, D.C. The protection that was required by existing operators necessitated both Dielectric antennas to have directional patterns pointed 1100 true, hardly ideal for either market, but with significant areas of signal overlap for SFN testing, as shown in Figure 30.

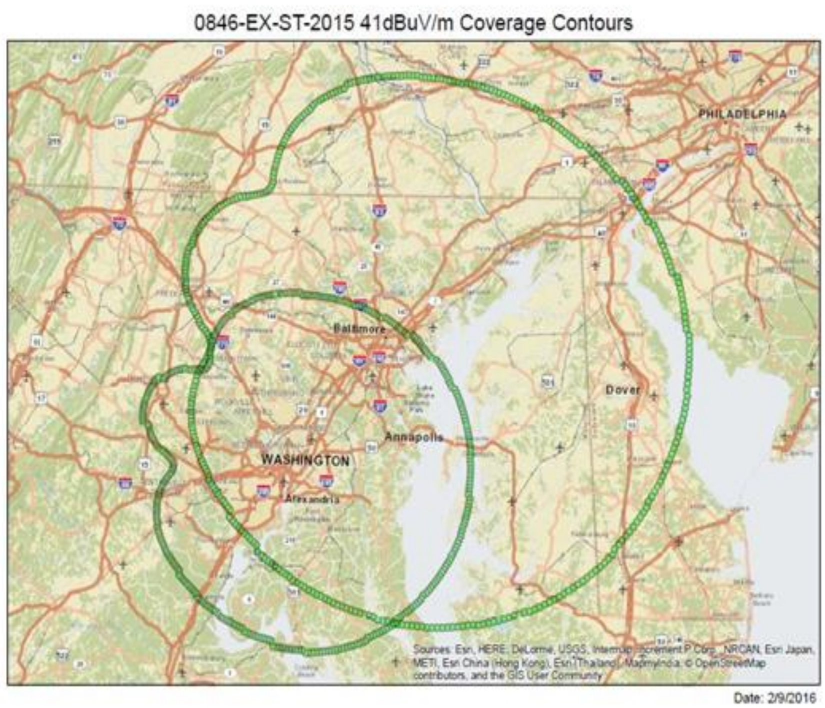

Figure 30: Antenna directional patterns, Baltimore/Washington D.C. field testing

Neither site had available equipment. As a result, two TV transmission facilities had to be built from scratch after the STA was awarded. The Baltimore site was planned as the typical tall tower, high-power site, widespread throughout the U.S. – 800kW ERP, 1,000’AGL, two-tube IOT transmitter running at 40kW TPO provided by Acrodyne Services. The Washington, D.C. site was designed according to what may become a typical SFN site with 120kW ERP, 350’AGL, solid-state transmitter at 6kW TPO provided by GatesAir. Both transmitters have Dielectric RF systems, including filters, transmission lines and antennas. The EXACT- ATSC 3.0 exciters for both sites were provided by Hitachi-Comark, and the ATSC 3.0 Modulator/Gateway Stream4Cast was provided by TeamCast. The Stream4Cast is located at the WBFF studio in Baltimore, and the modulated signal is distributed to each transmitter using a broadband private point-to-point Ethernet link.

Both sites were installed through the winter and were commissioned in late March, coincidentally and fortunately with the ATSC 3.0 PlugFest being held in Baltimore. This meant that ATSC 3.0-compliant receivers were available. The off-air reception of the SFN was verified, and a remote adjustment of the timing parameters was made with LG receivers.

An extension of the STA has been granted, and the intent is to carry out further field tests of signal reception when a suitable receiver is made available. The simple SFN has been modeled in ProgiraPlan, and field data will be fed back into the software for validation and optimization purposes. In the meantime, the plans are to characterize transmitter performance with ATSC 3.0 signals.

ADDENDUM B (ACRONYMS AND GLOSSARY)

2K - a scanning system that is 1080 lines by 1920 horizontal pixels.

3D (3 Dimensional) - in television, accomplished by encoding additional data such that the viewer can experience the perception of depth in the image. Images meant for viewing with one eye are encoded separately than the other. Images closer to the camera will be more diverged than imagery that is further away. This creates for the viewer an appearance of depth, if each image can be directed to the appropriate eye (as in through the use of special glasses).

4:2:2 - because the human visual system is less sensitive to the color of objects than their luminance component, chroma subsampling can be utilized while encoding video. In this specific case the sampling ratio of Y'CbCr is 4 to 2 to 2.

4:2:0 - in this scheme the chroma subsampling in each horizontal line is at half the luma rate, while Cb and Cr channels are only sampled on alternative lines.

4:4:4 - in this case the three Y'CbCr components have the same sample rate, thus there is no chroma subsampling required.

4K - a screen scan size that is 2160 lines by 3840 horizontal pixels (4 times the area of 1080 X 1920)

5.1 Channels - in ATSC-1, AC-3 audio describes 6 channels made up of Left, Center, Right, Left Rear, Right Rear and LFE (subwoofer).

7.1 Channels - in ATSC 3.0, the Next Generation Audio system will contain 7.1 channels which will include those mentioned in 5.1, as well as splitting the surround and rear channel information into four distinct channels, in which sound effects are directed to left and right surround channels, plus two rear surround channels.

8-VSB (8-Vestigial Side-Band) - the single-carrier waveform defined by the ATSC-1 standard and adopted by the FCC use as a digital television transmission standard in 1995.

A/53 - the primary ATSC document governing the ATSC-1 standard.

A/153 - the primary ATSC document governing the ATSC-M/H (mobile/handheld) standard.

AC-3 (Advanced Coding-3) - the descriptor for the audio codec developed by Dolby Labs and used in ATSC-1.

AC-4 (Advanced Coding-4) - the audio compression technology developed by Dolby Labs and used by convention in the US in ATSC-3.

ACE (Active Constellation Extension) - a methodology for reducing the negative effects of peak to average power ratio in COFDM transmitters, by use of manipulating the constellation to reduce the transmission Peak to Average Power Ratio.

AEA (Advanced Emergency Alerting) - a series of features that will be supported in ATSC that will bring vital emergency information to the viewer in the form of rich media, as well as other content that will be of vital interest to the viewer.

AFD (Active Field Descriptor) - a standard set of codes that can be sent in a video stream or in the baseband video signal that carries information about the desired display aspect ratio, as well as the screen rendering characteristics required. It is described in the SMPTE -2016-1 standard.

AGWN (Additive Gaussian White Noise) - a basic noise model that adds to the basic system noise model the effect of many random processes that occur in nature.

ALP (ATSC Link-layer Protocol) - the data encapsulation/compression abstraction layer used to provide baseband framing functions and signal input formatting.

ATSC - Advanced Television Standards Committee, an international industry-supported Standards Developing Organization (SDO) that develops and documents television transmission standards.

ATSC-1 - the first digital television transmission standard developed in 1995 that is used currently as the television standard in the U.S. as well as other countries.

ATSC-3 - the transmission standard being currently developed, that provides increased services with robust signal quality through the use of increased bandwidth efficiency and improved error detection and correction.

AVC (Advanced Video Coding) - MPEG4 Main 10 profile or H.264 video codec. It provides approximately twice the data bandwidth efficiency of MPEG-2 video encoding.

Base Layer - also referred to as Layer 1 of a two-layer LDM (Layered Division Multiplexing) system.

BBP (Base Band Packet) - a packet structure that will allow ALP packets to be encapsulated in such a way as to efficiently fit into the associated PLP structure. These packets contain padding or concatenation data as required**.**

BCH (Bose, Chaudhuri, Hocquenghem) - one of two options for linear error coding used in the BICM processing block for outer code correction (CRC is the other). For ATSC 3.0, a 12-bit correctable BCH provides for both error detection and correction capabilities.

BICM (Bit-Interleaved Coded Modulation) - a flexible modulation/coding scheme in which you may choose a modulation constellation independently of the coding rate. It contains the Forward Error Correction, Bit Interleaving and Mapping (as in constellation) functions.

Bit Interleaver - rearranges data without loss to provide immunity to time-based noise bursts that interfere with the received signal.

BMFF (Base Media File Format) - (ISO/IEC 14496-12 - MPEG-4 Part 12), a general structure for time-based multimedia files such as video and audio. It is designed as a flexible, extensible format that facilitates interchange, management, editing and presentation of the media, and is designed to be independent of any particular network protocol.

Bootstrap - also known as the System Discovery and Signaling (A/321), and is the universal receiver discovery signal into the ATSC-3 digital transmission signal. It precedes the preamble and is part of the overall ATSC 3.0 frame structure. In addition to the emergency alert wakeup signal, it includes ATSC 3.0 version number, and signals the FFT size, guard interval, and scattered pilot pattern of the preamble symbols.

Broadcast Gateway - a device that resides at the studio or NOC that provides IP delivery, as well as signaling, ALP processing, SFN processing, and scheduler functions.

Broadcast Manager - in a fully automated ATSC-3 transmission system, provides coordinated media preparation and delivery administration across content supply with over-the-air as well as broadband delivery.

Captioning (Closed) - data that is contained with the ATSC-3 signal that will provide a transcript version of what is being said on the screen. At the option of the viewer, the text may be displayed contemporaneously on the screen.

CAT 5 - a wiring standard described in TIA/EIA-568-B.1-2001 T568A or B with 100 Mhz in bandwidth, which translates into 1000BaseT.

CAT 6 - a wiring standard described in TIA/EIA-568-B.1-2001 T568A or B with 250 Mhz in bandwidth, which translates into 10GBaseT.

CFF-TT (Common File Format Timed Text) - a closed captioning data delivery standard based on W3C TTML with SMPTE-TT extensions.

C/N (Carrier to Noise) - a ratio, usually expressed in decibels (dB), of the carrier power relative to the noise power. This term is used to distinguish the C/N of the RF signal from the S/N of received data.

Code Rate - in a digital system with Forward Error Correction, this is the ratio of useful data to total data with redundancies (correction) included. For ATSC 3.0 there are 12 different code rates available (2/15 through 13/15)

COFDM (Coded Orthogonal Frequency Division Multiplex) – a digital multi-carrier modulation method that uses a large number of closely spaced carriers, 90 degrees apart, that carry complex data that has been converted from the frequency to the time domain.

Constellation - a two-dimensional visualization of a transmitted symbol of a complex digital number via the modulation of a sine and cosine signal.

Core Layer - the basic layer of an LDM transmission system.

CP (Circular Polarization) - a circularly polarized transmitted wave from an antenna which occurs when its vertical and horizontal components are equal. The relative phase of each component determines whether the result is right circular or left circular.

CRC (Cyclic Redundancy Check) - one of three options for the error coding used in the BICM processing block for outer code correction (BCH and none are the other two options). For ATSC 3.0, a 32-bit CRC provides only error detection with no error correction capabilities.

CTI (Convolutional Time Interleaver) - a means by which the data is pseudo-randomized to reduce the negative effects of random noise bursts in a transmission system. It is enabled when there is only a single PLP or when LDM is used with a single core-layer PLP.

DASH (Dynamic Adaptive Streaming over Hyper-Text Transfer Protocol) - in ATSC 3.0, an abstraction layer described in ISO/IEC 23009-1. It is a standard method to stream packets of (typically) video and audio over IP. The DASH format is used for both broadband and broadcast delivery. HTTP is used as the broadband delivery protocol, and ROUTE is used as its broadcast delivery protocol.

DECE (Digital Entertainment Content Ecosystem or Ultraviolet) - a consortium of content producers, consumer electronics manufacturers and Digital Rights Management (DRM) vendors that have created a series of standards for delivery of content.

Doppler –the often-detrimental phase and frequency shift that results from the relative motion of the receiver and transmitter. In a phase dependent modulation system such as COFDM it must be considered in the system design and error correction required. It is named for the scientist Christian Doppler who first discovered this phenomenon**.**

DP0, DP1, DP2 (Data Pipe 0, 1, 2) - three simultaneous time-division-multiplexed signals (or "data pipes") with an OFDM waveform. This terminology was used in the LG-Zenith-GatesAir complete system proposal for ATSC 3.0. This system was referred to as "Futurecast" by its proponents. This terminology is synonymous with the "PLP" term used in both DVB-T2 and ATSC 3.0. "PLP" means "Physical Layer Pipe."

DRM (Digital Rights Management) - a system by which digital media content is protected from unauthorized view or use (as in unauthorized copying).

DS3 (Digital Signal 3) - a commercially available digital data communications line. The data rate for a DS3 is 44.736 Mbit/s (45 Mb). A DS3 is alternatively known as a T3 Line.

DTV - Digital Television.

DTT (Digital Terrestrial Television) - television transmission system using digital transmission for broadcast, that makes efficient use of spectrum with the provision of more capacity than analog.

DVB-T2 (Digital Video Broadcasting, Second Generation Terrestrial) **-** the extension of the television standard DVB-T devised for the broadcast transmission of digital terrestrial television. DVB has been standardized by ETSI (*European Telecommunications Standards Institute)***.**

EAS (Emergency Alert System) - a national public warning system that requires broadcasters, cable television systems, wireless cable systems, satellite digital audio radio service (SDARS) providers, as well as direct broadcast satellite (DBS) providers to provide the communications capability to the President to address the American public during a national emergency. The system also may be used by state and local authorities to deliver important emergency information, such as AMBER alerts and severe weather information targeted to specific areas**.**

EBU (European Broadcast Union) - an alliance of public service media. It has 73 Members in 56 countries in Europe, and an additional 34 Associates in Asia, Africa and the Americas, broadcasting in more than 120 languages.

ECC (Error Correction Coding) – coding that uses an algorithm for expressing a sequence of bits such that any errors which are introduced in the system can be detected and corrected (within certain limitations) based on the remaining (redundant) bits.

EMDF (Extensible Metadata Delivery Format) – a protocol which provides a structured and extensible container for metadata to be carried in AC-4 bit streams.

Enhancement Layer - Layer 2 of a two-layer LDM system.

ERP (Effective Radiated Power) - the amount of power provided to a transmission line and antenna system minus the system losses and times the antenna gain. This is also the power level that the FCC authorizes for a broadcast facility.

ESG (Electronic Service Guide) - in ATSC 3.0, a file, likely delivered in non-real-time, that informs the viewer with a graphical guide about the contents of services available at any time, as well as how to access to those services.

EVM (Error Vector Magnitude) (sometimes also called Receive Constellation Error or RCE) - a measure (expressed in dB or %) of how far the transmitted and received constellation points are from the ideal locations.

Exciter/Modulator - in an ATSC 3.0 television transmitter, the functional block that contains the Input Formatter, Bit Interleaving and Coding, Framing and Interleaving as well as Waveform Generation.

FDM (Frequency Division Multiplexing) or OFMD (Orthogonal Frequency Division Multiplexing) - a modulation scheme that divides and carries a single digital signal (or its components) across thousands of signal carriers simultaneously. The OFDM carriers are sent at right angles to each other (hence, orthogonal) so they do not interfere with each other.

FEC (Forward Error Correction) - the process whereby additional (redundant) bits are added to a digital transmission that allows a receiver to detect bit errors and correct the signal using the redundant data.

FEL (Future Extension Layer) - an extension layer for an LDM system.

FFT (Fast Fourier Transform) **-** a process that mathematically converts a signal from its original time domain to a representation in the frequency domain.

FPGA (Field-Programmable Gate Array) - a high-density general-purpose integrated circuit that can be programmed to achieve specific signal processing tasks.

Frame - a data construct that includes a sequence of bits or symbols that indicate to the receiver the beginning and end of payload data.

FTP (File Transfer Protocol) - a standard network protocol used to transfer files between a client and server.

Futureproof - a system designed in such a way that it is unlikely to become obsolete.

Geolocation - a process or technique of identifying the geographical device location of a viewer by means of digital information, using various means such as GPS location or IP address (in the case of broadband connection).

GI (Guard Interval) - used to introduce immunity to propagation delays, echoes, and reflections. ATSC 3.0 has 12 user selectable GI lengths (192, 384, 512, 768, 1024, 1536, 2048, 2432, 3072, 3648, 4096, and 4864).

GUI (Graphical User Interface) - a type of user interface that allows users to interact with electronic devices through the use of graphical icons and visual indicators.

H.264 (also known as AVC or MPEG-4 Part 10, Advanced Video Coding) - a block-oriented motion-compensation-based video compression standard that is currently one of the most commonly used formats for the distribution of video content, that provides about twice the data bandwidth efficiency of MPEG-2.

H.265 (also known as High Efficiency Video Coding (HEVC) or MPEG-H Part 2) - a block-oriented motion-compensation-based video compression standard that is one of several potential successors to the widely-used H.264 or MPEG-4 Part 10, while providing nearly twice the bandwidth efficiency.

HDR (High Dynamic Range) - a technique used in video imaging to reproduce a greater dynamic range of luminosity than is possible with more standard digital imaging techniques or displays.

HD-SDI (High-Definition Serial Digital Interface) - the common method for high-definition digital video production and studio transmission of Y'CbCr component content, and is described in standard SMPTE 292M at a nominal data rate of 1.485 Gb/s.

HEVC (High Efficiency Coding) (also known as H.265 or MPEG-H Part 2) - a block-oriented motion-compensation-based video compression standard that is one of several potential successors to the widely-used H.264 or MPEG-4 Part 10, while providing nearly twice the bandwidth efficiency.

HFR (High Frame Rate) - television frame rates above the nominal rates of 60 frames in the U.S. and 50 frames in many other parts of the world. The higher frame rates would be 120 Hz (U.S.) or 100 Hz (elsewhere).

Hpol (Horizontal Polarization) - when an antenna has its electric field transmitted in the horizontal plane and the magnetic field in vertical plane.

HTI (Hybrid Time Interleaver) - a means by which the data is pseudo-randomized to reduce the negative effects of random noise bursts in a transmission system that utilizes the multiple-PLP mode. It is composed of cell interleaver, twisted block interleaver, and a convolutional delay-line.

HTTP (HyperText Transport Protocol) - an application or protocol for distributing hypermedia information using hyperlinks (addresses) to link from one hypertext file to another location or file.

HVAC (Heating Ventilation and Cooling) - the technology of controlling indoor environmental qualities of temperature and humidity.

Hybrid Service - in ATSC 3.0, a capability to make use of simultaneous over-the-air broadcast as well as delivery and return channel content via the internet. The internet-delivered content would presumably augment or be in addition to the over-the-air content.

IFFT (Inverse Fast Fourier Transform) - the process that mathematically converts a signal from its original frequency domain to a representation in the time domain. IFFT takes place in the waveform generation processing block of the ATSC 3.0 exciter/modulator.

Immersive (as in audio) - provides a realistic representation of the original sound field that appears to surround the user. Often referred to as theatre-quality sound.

IMSC1 (Internet Media Subtitles and Captions Version 1 or MSC-1) - the W3C standard on which ATSC 3.0’s caption and subtitle system is built.

Interlace – in television, the form of scanning in which an image is fully horizontally scanned at one half the frame rate, and alternately fully scanned again between the original scan locations, making up the full frame.

Internet Protocol (IP) - the digital protocol by which data is sent from one device to another via the internet or a network. Each source of data has at least one or more IP addresses that uniquely identifies it from all other data sources. Destination devices often have IP addresses as well to be uniquely identified or addressed. However, the protocol also makes provision for “broadcast” data in which only the source address is required.

IOT (Inductive Output Tube) - a type of high power linear beam vacuum tube that uses current modulation that is primarily used in UHF transmitters. Developed in the 1980s, IOTs provide an alternative technology to klystrons, providing greater efficiency and lower operating costs.

IP V4 (Internet Protocol Version 4) - the fourth revision of the Internet Protocol (IP) definition, and a widely-used protocol in data communication over different types of networks.

IP V6 (Internet Protocol Version 6) - the sixth revision of the Internet Protocol (IP) definition, and a widely-used protocol in data communication over different types of networks. IP V6 is the enhanced version of IP V4, and can support very large numbers of nodes as compared to IP V4.

IPTV (Internet Protocol Television) - a system through which television services are delivered using Internet Protocol over packet-switched networks, as in the internet.

ISDB-T (Integrated Services Digital Broadcasting, Terrestrial) - a Japanese standard for digital television. ISDB-T replaced the previously used MUSE Hi-vision analog HDTV system.

ISO/BMFF (ISO/IEC 14496-12 - MPEG-4 Part 12) - a general structure for time-based multimedia files such as video and audio. It is designed as a flexible, extensible format that facilitates interchange, management, editing and presentation of the media. It is designed to be independent of any particular network protocol.

ISO/IEC 23008-1 - specifies MPEG Media Transport (MMT) technologies, which include a single encapsulation format, delivery protocols and signaling messages for transport and delivery of multimedia data over packet-switched networks.

ITU Rec. 709 (also known as BT.709) - standardizes the format of high-definition television, having 16:9 (widescreen) aspect ratio with a defined transfer function and color space definition.

ITU Rec. 2020 (also known as BT2020) - defines various aspects of HDTV such as improved display resolution, frame rate, Chroma subsampling, bit depth, and color space over ITU Rec. 709.

L1 Basic - part of the Preamble following the “bootstrap,” and carries the most fundamental signaling information as well as data necessary to decode L1 Detail.

L1 Detail - part of the Preamble following the L1 Basic. It has the information necessary to decode subframes including their ModCods, number of PLPs, pilot pattern, FEC, etc.

Layer - a conceptual model that characterizes and standardizes the communication functions of a data system while isolating it from the technology utilized. Such a model partitions the system into abstraction (independent) layers.

LCT (Layer Coding Transport) (also known as RFC 5651) - provides transport level support for content delivery and stream delivery protocols such as ROUTE/DASH or ROUTE/UDP/IP. LCT is specifically designed to support protocols using IP multicast, but it also provides support to protocols that use unicast.

LDM (Layered Division Multiplexing) - a multiplexing scheme where multiple RF signals are layered on top of one another. A two-layer system has a core layer (more robust ModCod) and an enhanced layer (less robust ModCod). The enhanced layer is “injected” between -3 and -10dB relative to the core layer.

LDPC (Low-Density Parity Check) **-** a linear error correcting code, used in the BICM processing block for inner code correction. Inner code correction is mandatory in ATSC 3.0. There are two different sizes of the LDPC code: 64800 bits and 16200 bits.

Lighthouse Station - a method by which, during the industry transition to ATSC 3.0, multiple stations in a market will transmit ATSC 3.0 services on a single designated channel, using separate PLPs or stamux on a single PLP. This would facilitate a transition in a market because viewers could still view the stations' ATSC-1 transmissions while transitioning to ATSC-3.

LLS (Low Level Signaling) - signaling information that supports rapid channel scans and bootstrapping of service acquisition by the receiver. It operates below the IP layer, and includes a table that points to the Service List Table (SLT), Regional Ratings Table (RRT), System Time (ST), Common Alerting Protocol (CAP), and Service Layer Signaling (SLS) tables.

LMT (Link Mapping Table) - provides a table or list of the upper layer sessions carried in a PLP.

MIMO (Multiple Input Multiple Output) - one of three frame types (SISO, MISO, MIMO). MIMO improves system robustness via additional spatial diversity (two transmit, two receive antennas). The spatial diversity is often combined with polarization diversity (Hpol and Vpol).

MISO (Multiple Input Single Output) - one of three frame types (SISO, MISO, MIMO). MISO is a pre-distortion technique that artificially de-correlates signals from multiple transmitters in a Single Frequency Network in order to minimize potential destructive interference.

MMTP (Multi Media Transport Protocol) - an application layer transport protocol for delivering multimedia streams over IP networks

ModCod (Modulation and Code Rate) - the combination of modulation and coding rates that together determine the size of the baseband packet.

MPEG-2 TS (Motion Picture Expert Group-2 Transport Stream) **-**a digital container format for streaming television signals that is part of the MPEG-2 Part 1 specification.

MPEG-H (Motion Picture Experts Group-High efficiency coding and media delivery in heterogeneous environments) - a group of standards that includes next-generation audio and video compression technologies.

MSC1 (Internet Media Subtitles and Captions Version 1 or IMSC-1) - the W3C standard on which ATSC 3.0's caption and subtitle system is built.

Network Layer Packet - an elemental Packet Structure that provides payload content along with its routing information.

NGA (Next Generation Audio) - audio provided in a highly efficient digitally compressed format that delivers immersive quality audio, along with a host of features such as customized channel selection control.

Nightlight Station - a concept by which, when the ATSC-3 transition is fairly mature and all of the stations have transitioned to ATSC-3, a single station transmits a multiplex of all of the stations in the market in ATSC-1 as to not orphan legacy receiver viewers.

NOC (Network Operations Center) - the facility that contains the system resource manager, data sources/program encoders, and the broadcast gateway. The NOC may also be called master control in many facilities.

NRT (Non Real Time) - in ATSC 3.0, the concept of delivering file content or applications non-contemporaneously (generally before time) with their intended use.

NUC (Non-Uniform Constellation) - a constellation (QAM) with a non-uniform spread of the constellation points. Such constellations provide additional shaping gain, which allows reception at lower signal-to-noise ratios.

Null Fill - an antenna phasing design used to fill in signal coverage in areas of the pattern that do not have desired signal strength.

OC3 (Optical Carrier 3) - a network line with a transmission data rate of up to 155.52 Mbit/s using primarily fiber optics. OC-3 may also be known as STS-3 or STM-1.

OFDM (Orthogonal Frequency Division Multiplexing) - a digital multi-carrier modulation method that uses a large number of closely spaced carriers, 90 degrees apart, that are used to carry complex data that has been converted from the frequency to the time domain.

Offloading - where data, video and other bandwidth-intensive content can be transmitted over broadcast networks for “edge” storage or delivery to non-household destinations, freeing required bandwidth for other uses.

OSI (7 Layer Model) - the model which defines a networking framework to implement protocols in seven layers. Those layers are Physical (Layer 1), Data Link (Layer 2), Network (Layer 3), Transport (Layer 4), Session (Layer 5), Presentation (Layer 6), and Application (Layer 7). Each layer is an abstraction (independent) layer that provides for extensibility by not relying on the characteristics of other layers. This allows for the separation of requirements to facilitate interoperability and platform independence. ATSC 3.0 is built on just such an architecture.

OTA (Over-The-Air) - programs directly received from a local transmission.

OTT (Over-The-Top) - television programming, streams or multimedia services received via methods other than over-the-air, without the involvement of a multiple-system operator in the control or distribution of the content. The term originated from the concept of receiving the streams “over-the-top” of cable television broadband connectivity. However, the term has since broadened to include television delivery via internet broadband in general.

PAPR (Peak-to-Average Power Reduction or PAR) - a transmitter’s peak power squared divided by the average (RMS) power squared, and is expressed generally in dB. It is also equivalent to crest factor.

PAPR Reduction - modifies the ODFM signal via Tone Reservation (TR) and/or Active Constellation Extension (ACE) to reduce the peak power requirements of the ATSC 3.0 transmission.

PLP (Physical Layer Pipe) - a logical data transmission channel that may carry one or multiple services. Each PLP can have different bit rate and error protection parameters. It provides a data and transmission structure of allocated capacity and robustness that can be adjusted to broadcaster needs. In ATSC 3.0, the maximum number of PLPs in an RF channel is 64. Each individual service can utilize up to 4 PLPs. Therefore, receivers are expected to be able to decode at least four PLPs at one time.

PNG (Portable Network Graphics) - a raster graphics file format that supports lossless data compression. PNG is the most commonly used lossless image compression format on the internet.

Preamble - present at the beginning of an ATSC 3.0 frame or group of subframes. It contains the Level 1 control signaling applicable to the remainder of the frame(s). The preamble has two parts: L1 Basic and L1 Detail.

Progressive - in television, a scanning method that scans the frame completely in one pass.

PSIP (Program System Information Protocol) - in ATSC 1.0, a collection of tables describing virtual channel attributes, event features, and other information. The complete specification is described in ATSC standard A/65.

QAM (Quadrature Amplitude Modulation) - a signal in which two carriers shifted in phase by 90 degrees are modulated, summed and the resultant output consists of both amplitude and phase variations. In the ATSC 3.0 physical layer, constellations resulting from QAM modulation range by broadcaster choice from QPSK to 4096QAM. High spectral efficiencies can be achieved with QAM by setting a suitable constellation size, limited only by the noise level and linearity of the channel required.

QPSK (Quadrature Phase Shift Keying) - a digital modulated signal consisting of a two bit (4 point, or quadrature) QAM constellation that is usually used for low bit rate, high robust transmission.

RAP (Random Access Point) - a randomly selected (non-sequential) location in a digital signal that is used as a reference location for synchronizing a process.

Repack - will be the ultimate result of the FCC's spectrum incentive reverse-auction in 2016/2017 to buy spectrum from broadcasters that will, in turn, be sold to wireless operators in a forward auction. With the resultant consolidated spectrum set aside for television broadcast, some stations will need to move to a different part of the band to clear the sold spectrum for wireless use.

Return Channel - in ATSC 3.0, a data transmission link from a viewer's receiver back to the broadcaster's facility. The return channel in ATSC 3.0 may use the internet or an RF transmission channel.

ROI (Return on Investment) - the amount of monetary return relative to the investment's cost.

ROUTE (Real-time Object delivery over Unidirectional Transport) - an IP-centric transport protocol that is compatible with layered environments and is based on IETF protocols. In ATSC 3.0, it is used to carry a DASH session of multimedia content.

RTP (Real-time Protocol) - a network protocol for delivering audio and video over IP networks. RTP is used extensively in communication and entertainment systems that involve streaming media, and is described in IETF RFC-3550.

Scheduler - a functional processing block within the Broadcast Gateway, at the master control or NOC, that allocates physical capacity for the services required by the broadcaster in ATSC 3.0 transmissions.

Service - a set of content elements, when taken together, which provide a complete listening, viewing, or other experience to a viewer. It may contain audio, base level video, enhancement video, captioning, graphic overlays, web pages, applications, emergency alerts as well as other signaling or metadata required.

Service Guide - in ATSC 3.0, a file, likely delivered in non-real-time, that informs the viewer in a graphical manner about the contents of services available at any time, as well as how to access those services.

SFN (Single Frequency Network) - two or more transmitters operating on the same channel in a synchronized manner, generally to improve transmission coverage.

SISO (Single Input Single Output) - one of three frame types (SISO, MISO, MIMO). SISO is signal processing with only one transmit antenna and only one receive antenna required for full reception.

SLS (Service Layer Signaling) - provides to the receiver sufficient information to discover and access ATSC 3.0 services and their content components.

SLT (Service List Table) - in ATSC, it enables the ATSC 3.0 receiver to build a basic service list while pointing to the location of the SLS (Service Layer Signaling).

SMPTE 2016-1- the SMPTE standard for Active Field Descriptor, which is a standard set of codes that can be sent in a video stream or in the baseband video signal that carries information about the aspect ratio, as well as the screen rendering characteristics required.

SMPTE 2022-1- Forward Error Correction for Real-Time Video/Audio Transport over IP Networks. It also defines row/column FEC (Forward Error Correction) for IP video streams. The row/column FEC works by grouping IP video packets into logical rows and columns, and then appends one FEC packet to each row and each column.

SNR (Signal to Noise Ratio) - compares the level of a desired signal to the level of background noise. It is defined as the ratio of signal power to the noise power, often expressed in decibels. In digital communication systems, quantization errors are a common source of the noise.

SMPTE-TT - defines the SMPTE profile of W3C Timed Text Markup Language (TTML) used to transmit Captions or Subtitles. It identifies the features from TTML required for interoperability between display systems for the format. SMPTE-TT also defines some standard metadata terms to be used, and some extension features not found in TTML.

Soundbar - a single cabinet speaker system with a small footprint, built with small phased speakers that can simulate surround-sound.

Spatial - in video encoding, those items, errors or corrections that occur within a frame.

Spectrum Repack - will be the result of the FCC's spectrum incentive reverse auction in 2016/2017 to buy spectrum from television stations, then sell the purchased and consolidated spectrum to wireless providers. When this process is complete, some television stations will need to move to a different part of the band to clear this spectrum for the wireless use.

SSC (Spatial Scalable Coding) - enables the encoding of a high-quality video bitstream that contains one or more subset bitstreams that can themselves be decoded with a reconstruction quality somewhat similar to that of the original bitstream. The subset bitstream is derived by dropping packets from the larger bitstream. The subset bitstream can represent a lower spatial resolution (smaller screen), or a lower temporal resolution (lower frame rate), compared to the original bitstream.

STA (Special Temporary Authority) - under FCC rules, provides for immediate operation for broadcast station's transmission when temporary authority is required because licensed facilities have been damaged or experimental transmission is requested.

STL (Studio to Transmitter Link) - the transmission link between the broadcaster's studio location and the transmitter, carrying the station's content to be transmitted. This link may be via radio means (microwave) or via direct digital connection, such as fiber.

STLTP (Studio to Transmitter Transport Protocol) - In ATSC 3.0, provides a STL transmission interface between the Broadcast Gateway, located at the studio, and the transmitter(s) exciter/modulator. It encapsulates payload data using UDP, provides synchronization time data and control, as well as STL forward error correction.

Subframe - in ATSC 3.0, a PLP may contain a structure of a frame or a series of subframes. Each subframe may have separate transmission characteristics. There is a bootstrap sequence and preamble is found at the beginning of each frame or series of subframes.

Sweetspot - in multichannel audio, describes the focal point between multiple speakers, where an individual is fully capable of hearing the stereo audio mix in a way it was intended to be heard.

TCP/IP (Transport Control Protocol via Internet Protocol) - the basic communication language or protocol of the internet or other IP-based delivery systems, as in a private network. It requires two-direction (duplex) connectivity.

TDM (Time Domain Multiplex) - a method of joining multiple data streams into a single stream by dividing the source streams into many timed segments, each of short time duration, and interleaving them into the common stream. The individual data streams can then be reassembled at the receiving end by reversing the process, based on the timed segment duration.

Temporal - in video encoding, those items, errors or corrections that occur between frames.

Tone Reservation - in COFDM transmission, a method for reducing Peak to Average Power by adding (reserving) subcarriers (tones) that don't carry any data information, for the purpose of reducing PAPR.

TPO (Transmitter Power Output) - the actual amount of RF power that a transmitter produces at its output connection.

Transcript File - a transcription or translation of the dialogue text, sound effects, relevant musical cues, and other relevant audio information in text form, used to create a closed captioning file.

Transfer Function - in television, is used to mathematically describe what the response of an optical sensor is to a wide range of light levels. There is rarely a linear light-to-signal output relationship, so for the imagery to be properly rendered, the display device must emulate the inverse transfer function.

TTML (Timed Text Markup Language) - a W3C-developed closed-captioning data-delivery standard. CFF-TT (Common File Format Timed Text) is based on TTML with SMPTE-TT extensions.

UDP (User Datagram Protocol) - a data delivery standard, defined by RFC 768, that delivers its payload as datagrams *(*header and payload sections) to devices on an IP network. UDP provides checksums for data integrity, and port numbers for addressing different functions. There are no handshaking dialogues, and thus UDP can be used in single-direction communications.

UHD (Ultra High Definition) - a descriptor of the highest technical quality for television programming, which includes 4K resolution, high display refresh rate, High Dynamic Range, Wide Color Gamut, and immersive audio.

VOD (Video On Demand) - an interactive TV technology that allows subscribers to view programming in real time, or download programs to view later.

Vpol (Vertical Polarization) - when an antenna has its electric field transmitted in the vertical plane and the magnetic field in the horizontal plane.

W3C (World Wide Web Consortium) - an international community where member organizations, a full-time staff, and the public work together to develop standards to be used on the World Wide Web.

WCR (Wide Color Gamut) - a wider range of color values that are closer to the human visual range than prior color descriptions. The television wide-gamut color space is defined in ITU Rec. 2020 that covers 75.8% of the visual color space. The color space currently used in television ITU Rec. 709 covers only 35.9% of the visual color space by comparison.

Y'CbCr - a family of color space used in video systems. Y' is the Luma component and Cb and Cr are the blue and red Chroma color difference signals. The prime on the "Y" is to distinguish Luma from Luminance. Luma differs from the scientific term Luminance, which does not have the gamma (transfer function) correction found in Luma as is used in television imagery.

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