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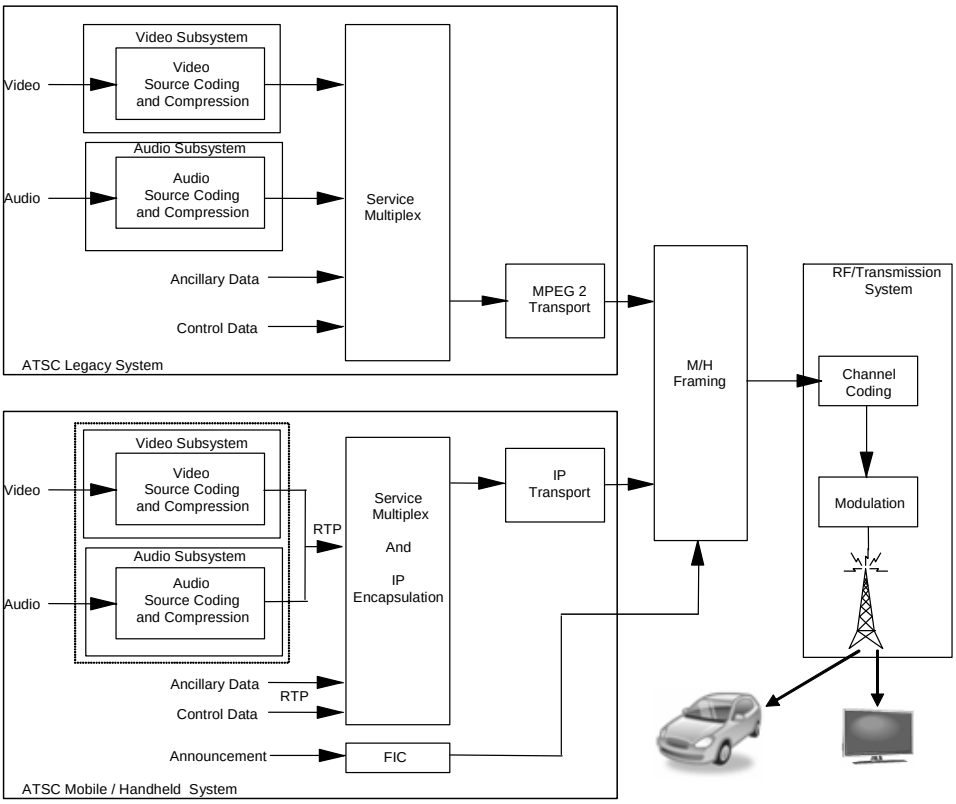
ATSC APPROVES MOBILE/HANDHELD DTV CANDIDATE STANDARD

On November 25, 2008, the ATSC's Technology and Standards Group (TSG) approved a new Candidate Standard for Mobile/Handheld (M/H) Digital Television. This technology will provide local broadcasters with a major opportunity for new services in their over-the-air digital channel. The standard results from an intense development effort within ATSC involving hundreds of participants around the world and across the television and mobile device industries. It started with a Call for Proposals for a Mobile and Handheld Standard – see *TV TechCheck* of May 21, 2007. Further details are as follows.

The ATSC M/H system is intended to provide mobile/pedestrian/handheld broadcasting services using a portion of the ~19.39 Mbps ATSC 8-VSB payload, while the remainder is still available for HD and/or multiple SD television services. In very simple terms, the M/H system achieves the robustness needed for mobile reception by adding extra training sequences and forward error correction. The total bandwidth needed for the M/H service depends on several factors, including the number and type of program services, the quality level, and level of robustness desired, typically ranging from less than one megabit per second to many megabits per second. The M/H system converts the current 8-VSB emission into a dual-stream system without altering the emitted spectral characteristics. It does this by selecting some of the MPEG-2 segments (corresponding to MPEG-2 Transport packets in the current system) and allocating the payloads in those segments to carry the M/H data in a manner that existing legacy receivers can ignore.

The figure to the right shows high level functional blocks for the current (legacy) ATSC system and the Mobile/Handheld system.

As well as providing improved reception capabilities, there are several major changes introduced in the ATSC-documented technology. For improved efficiency, the video is coded using the MPEG-4 Part 10 codec, generally known as Advanced Video Coding (AVC), with an option to use Scalable Video Coding (SVC), and audio compression uses MPEG-4 HE-AAC v2. Another major change is that the system transport uses Internet Protocol (IP) standards, as contrasted with the familiar 188-byte MEPG-2 Transport Stream. The system has provisions for service protection, interactive user



Block Diagram with Legacy ATSC and Mobile/Handheld Systems

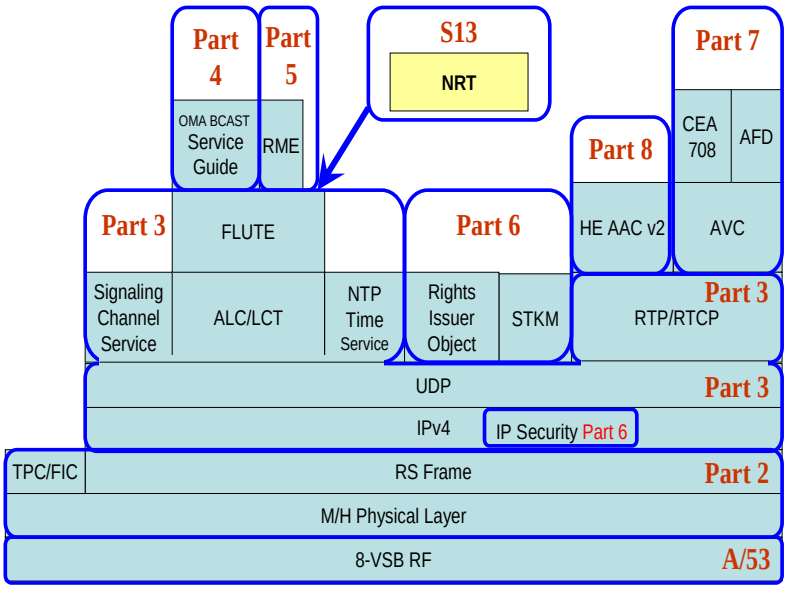
interfaces (one way and two way), a complete electronic service guide and non-real time data streams. Built-in ways are provided to enable further changes to occur, as technology develops, without causing malfunctions in M/H receivers. This provides forward compatibility for system upgrades and enables broadcasters to quickly react to marketplace challenges.

The A/153 Candidate Standard that documents the M/H system is organized in eight parts roughly corresponding to layers often used to model communication systems:

Part 1 is “Mobile/Handheld Digital Television System”
Part 2 is “RF/Transmission System Characteristics”
Part 3 is “Service Multiplex and Transport Subsystem Characteristics”
Part 4 is “Announcement”
Part 5 is “Application Framework”
Part 6 is “Service Protection”
Part 7 is “AVC and SVC Video System Characteristics”
Part 8 is “HE AAC v2 Audio System Characteristics”

The following figure shows how these layers are arranged in a stack, and indicates the parts that document the requirements.

Part 1 describes the M/H system overview and the organization of the standard. It also describes the explicit signaling requirements that are implemented by data structures throughout the other parts.



ATSC M/H System Protocol Stack and Associated Document Parts

Part 2 describes how the M/H data is processed and placed into the VSB frame. There are two major aspects, shown as two layers. Simplified, the M/H Physical layer can be thought of as the techniques to ‘hide’ this data from legacy receivers and assist equalizer convergence, and enable time slicing by receivers. The next layer has a Transmission Parameter Channel (TPC) and a Fast Information Channel (FIC) along with the Reed-Solomon (RS) Frame which has the Forward Error Correction (FEC) coded payload. The TPC signals the M/H transmission parameters that instruct the receiver how to recover the FIC and the RS frames. The FIC enables direct access to a selected RS frame and a selected service within the RS frame.

The M/H data in the RS frames is placed in bursts called “slots” in order to establish a time-slicing mechanism to enable receiver power management. A receiver can remain “off” until just before the next slot that has data for the selected service is about to be transmitted.

M/H data in these RS frames is organized into Ensembles, each of which contains one or more services. Each Ensemble uses an independent RS Frame (the FEC structure), and furthermore, each Ensemble may be coded to a different level of error protection depending on the application.

Part 3 covers the service multiplex and transport subsystem, which comprises several layers in the stack. The first layer uses Internet Protocol (v4), optionally with encryption of the contents of the IP packets containing services. As the system is required to function in the absence of a return channel the UniDirectional Protocol (UDP) is used. This enables multicast service joining in progress.

In general, there are two types of content that can be delivered using M/H. The first of these is streams of packets, such as television or audio programming. These are sent using Real Time Protocol (RTP). The second is packets containing files, sent using the one-way FLUTE over ALC/LCT internet protocol (the boxes over the box “FLUTE” in the figure). Files may contain navigation information; content to be stored and played later (non-real time programs), security objects and keys (LTKM) for service protection. The Signaling Channel Service used to identify the content being carried is similar to the Virtual Channel Table in PSIP.

Part 3 also contains a buffer model to enable control of audio-video synchronization over IP delivery using Network Time Protocol (NTP) in a similar manner to that used for current DTV services with a System Time Clock.

Part 4 covers Announcement. In the M/H system, Services can optionally be announced using a Service Guide. The guide used in the standard is based on an Open Mobile Alliance (OMA) broadcast (BCAST) Service Guide, with constraints and extensions. A Service Guide is delivered using one or more IP streams. The main stream delivers the Announcement Channel, and zero or more streams are used to deliver the guide data. If separate streams are not provided, guide data is carried in the Announcement Channel stream. The Service Guide is designed so that it may also be delivered over a separate connection if a device has two-way connectivity.

Part 5 defines the Application Framework. This sub-system enables the broadcaster of the audio-visual service to author and insert supplemental content to define and control various additional elements to be used in conjunction with the M/H audio-visual service. It enables the definition of auxiliary (graphical) components, layout for the service, transitions between layouts and composition of audio-visual components with auxiliary data components. Furthermore, it enables the broadcaster to send remote events to modify the presentation and to control the presentation timeline. The Application Framework further enables coherent rendering of the service and its layout over a variety of device classes and platforms, rendering of action buttons and input fields, and event handling and scripting associated with such buttons and fields.

Part 6 covers Service Protection, which refers to the protection of content, either files or streams, during its delivery to a receiver. Service Protection is an access control mechanism intended for subscription management. It assumes no responsibility for content after it has been delivered to the receiver.

Part 7 defines the AVC and SVC Video System. The M/H system uses MPEG-4 AVC and SVC video coding as described in ISO/IEC 14496 Part 10, with certain constraints. A single base format of 240 lines x 416 pixels, 16:9 aspect ratio, progressive scan, is specified, with the ability to increase the resolution or quality through use of the SVC option.

Part 8 defines the HE-AAC v2 Audio System. The M/H system uses MPEG-4 HE-AAC v2 audio coding as described in ISO/IEC 14496 Part 3, with certain constraints, and is used to code mono or stereo audio.

NRT – Non-Real-Time. While the basic capability for sending files is defined in Part 3, it is very general and broad. A constrained set of interoperability points are being defined in a separate ATSC activity (in TSG S/13) where non-real-time file delivery requirements are being documented for use in both the M/H system and the current 8-VSB system.

The M/H Candidate Standard will be available on the ATSC Web site at www.atsc.org in early January 2009. ATSC Candidate Standards are documentation of a technology that is mature enough to begin trial implementations in order to confirm that the documentation is complete and is an explicit call for technical feedback. Changes can be made as issues are reported. Once it is believed that no further substantive changes are needed, the CS is voted on by TSG to be promoted to a Proposed Standard, which, after full ATSC approval, becomes an approved Standard. The M/H CS is targeted to be promoted in May 2009. This gives the possibility for start of M/H services by broadcasters later in 2009.