HD Radio™ Air Interface Design Description Audio Transport

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1 Scope

1.1 System Overview

The iBiquity Digital Corporation HD Radio™ system is designed to permit a smooth evolution from current analog amplitude modulation (AM) and frequency modulation (FM) radio to a fully digital in-band on-channel (IBOC) system. This system delivers digital audio and data services to mobile, portable, and fixed receivers from terrestrial transmitters in the existing medium frequency (MF) and very high frequency (VHF) radio bands. Broadcasters may continue to transmit analog AM and FM simultaneously with the new, higher-quality and more robust digital signals, allowing themselves and their listeners to convert from analog to digital radio while maintaining their current frequency allocations.

1.2 Document Overview

This document describes the design and capabilities of the audio transport. It describes how control and information are passed through this transport. This document also details the requirements imposed on the audio codec by the design of the overall HD Radio system. Specific hardware and software implementations are is not described.

2 Referenced Documents

[1] iBiquity Digital Corporation, "HD Radio™ Air Interface Design Description - Layer 1 FM," Doc. No. SY_IDD_1011s, Revision G.

[2] iBiquity Digital Corporation, "HD Radio™ Air Interface Design Description - Layer 1 AM," Doc. No. SY_IDD_1012s, Revision F.

[3] iBiquity Digital Corporation, "HD Radio™ FM Transmission System Specifications," Doc. No. SY_SSS_1026s, Revision E.

[4] iBiquity Digital Corporation, "HD Radio™ AM Transmission System Specifications," Doc. No. SY_SSS_1082s, Revision E.

[5] iBiquity Digital Corporation, "HD Radio™ Air Interface Design Description – Layer 2 Channel Multiplex Protocol," Doc. No. SY_IDD_1014s, Revision H.

[6] iBiquity Digital Corporation, "HD Radio™ Air Interface Design Description – Advanced Application Services Transport," Doc. No. SY_IDD_1019s, Revision F.

[7] iBiquity Digital Corporation, "HD Radio™ Air Interface Design Description – Station Information Service Protocol," Doc. No. SY_IDD_1020s, Revision G.

[8] iBiquity Digital Corporation, "HD Radio™ Air Interface Design Description – Program Service Data," Doc. No. SY_IDD_1028s, Revision E.

[9] iBiquity Digital Corporation, "HD Radio™ Air Interface Design Description –Program Service Data Transport," Doc. No. SY_IDD_1085s, Revision C.

3 Abbreviations and Conventions

3.1 Abbreviations and Acronyms

AF	Audio Frame
AM	Amplitude Modulation
AAS	Advanced Application Services
C	Center
C/E	Core/Enhanced
CRC	Cyclic Redundancy Check
ECPL	Embedded Code PDU Length
FM	Frequency Modulation
GF	Galois Field
IBOC	In-Band On-Channel
ID	Identification
L	Left
MF	Medium Frequency
MPA	Main Program Service Audio
MPS	Main Program Service
N/A	Not Applicable
NOP	Number of Packets
PAD	Program Associated Data
PCI	Protocol Control Information
PDU	Protocol Data Unit
PCM	Pulse Code Modulation
R	Right
RBDS	Radio Broadcast Data System
RS	Reed-Solomon
RX	Receiver
SPS	Supplemental Program Service
VHF	Very High Frequency

3.2 Presentation Conventions

Unless otherwise noted, the following conventions apply to this document:

- All vectors are indexed starting with 0.
- The element of a vector with the lowest index is considered to be first.
- In drawings and tables, the leftmost bit is considered to occur first in time.
- Bit 0 of a byte or word is considered the least significant bit.
- When presenting the dimensions of a matrix, the number of rows is given first (e.g., an n x m matrix has n rows and m columns).
- In timing diagrams, earliest time is on the left.
- Binary numbers are presented with the most significant bit having the highest index.
- In representations of binary numbers, the least significant bit is on the right.

3.3 Mathematical Symbols

3.3.1 Variable Naming Conventions

The variable naming conventions defined below are used throughout this document.

Category	Definition	Examples
Lower and upper case letters	Indicates scalar quantities	i, j, J, g_{11}
Underlined lower and upper case letters	Indicates vectors	$\underline{u}$, $\underline{V}$
Double underlined lower and upper case letters	Indicates two-dimensional matrices	$\underline{\underline{u}}$, $\underline{\underline{V}}$
[i]	Indicates the i^{th} element of a vector, where i is a non-negative integer	$\underline{u}[0]$, $\underline{V}[1]$
[ ]	Indicates the contents of a vector	$\underline{v}$ = [0, 10, 6, 4]
[i] [j]	Indicates the element of a two-dimensional matrix in the i^{th} row and j^{th} column, where i and j are non-negative integers	$\underline{\underline{u}}[i][j]$, $\underline{\underline{V}}[i][j]$
$\begin{bmatrix} \quad & \quad \\ & \end{bmatrix}$	Indicates the contents of a matrix	$\underline{\underline{m}} = \begin{bmatrix} 0 & 3 & 1 \\ 2 & 7 & 5 \end{bmatrix}$
n …m	Indicates all the integers from n to m, inclusive	3 …6 = 3, 4, 5, 6
n:m	Indicates bit positions n through m of a binary sequence or vector	Given a binary vector i = [0, 1, 1, 0, 1, 1, 0, 0], $i_{2:5}$ = [1, 0, 1, 1]
NOP	No. of Packets	NOP=64

3.3.2 Arithmetic Operators

The arithmetic operators defined below are used throughout this document.

Category	Definition	Examples
·	Indicates a multiplication operation	3·4 = 12
INT( )	Indicates the integer portion of a real number	INT(5/3) = 1 INT(-1.8) = -1
a MOD b	Indicates a modulo operation	33 MOD 16 = 1
$\oplus$	Indicates modulo-2 binary addition	$1 \oplus 1 = 0$
|	Indicates the concatenation of two vectors	$\underline{A}$ = [$\underline{B}$ | $\underline{C}$] The resulting vector $\underline{A}$ consists of the elements of $\underline{B}$ followed by the elements of $\underline{C}$.
j	Indicates the square-root of -1	j = $\sqrt{-1}$
Re( )	Indicates the real component of a complex quantity	If x = (3 + j4), Re(x) = 3
Im( )	Indicates the imaginary component of a complex quantity	If x = (3 + j4), Im(x) = 4
$\log_{10}$	Indicates the base-10 logarithm	$\log_{10}(100) = 2$
ceil(numeric)	Smallest integer not less than argument	ceil(-42.8) = -42

4 Audio Transport – Detailed Design Description

This section describes the Audio Transport design, emphasizing its operations, processing, and interfacing of the audio encoder within the Audio Transport layer. The following broad system concepts are presented:

- Maintaining the fixed and variable encoding rates of the Audio Encoder
- Audio Transport delay control
- *Audio Frame* (AF) size and number
- Time alignment of analog and digital signals
- Data transport

Detailed audio encoder interface descriptions are organized in a functional sense to present guidelines for audio compression systems operating within the iBiquity HD Radio system.

All the bit rates mentioned in this document are "transport" rates which include the net codec rate and all applicable overhead. Also, the audio clock is derived from the broadcast system clock.

Note: All aspects of the Audio Transport design in this document also apply to the Supplemental Program Service (SPS) and the generation of SPS PDUs unless mentioned otherwise.

4.1 Introduction

Figure 4-1 shows the interface of the Audio Transport layer to the rest of HD Radio system. During broadcast, the Audio Encoder receives input audio frames from the Audio Interface, encodes the audio samples into Audio encoded packets, generates MPS PDUs in the Audio Transport, and conveys the MPS PDUs as output data streams to be transmitted. In addition, the Audio Transport obtains Program Service Data (PSD) byte-streams, if present, from the PSD Transport [9] and multiplexes this data with encoded audio. Thus, the output streams contain both compressed audio and PSD. PSD provides additional information about the audio program being transmitted; both the Main Program Service Audio (MPSA) and the Supplemental Program Service Audio (SPSA).

The Audio Encoder may generate one or two encoded audio streams (core and enhanced), depending on the audio codec mode as indicated by the dotted lines in Figure 4-1. Finally, the audio encoder indicates the amount of unused capacity to the Audio Transport, which relays the unused capacity status to the AAS Transport [6], thus allowing the inclusion of opportunistic data. The definitions of the units used for data transfer – audio frames, encoded audio packets, MPS PDUs – within the Audio Transport layer are explained in the Glossary.

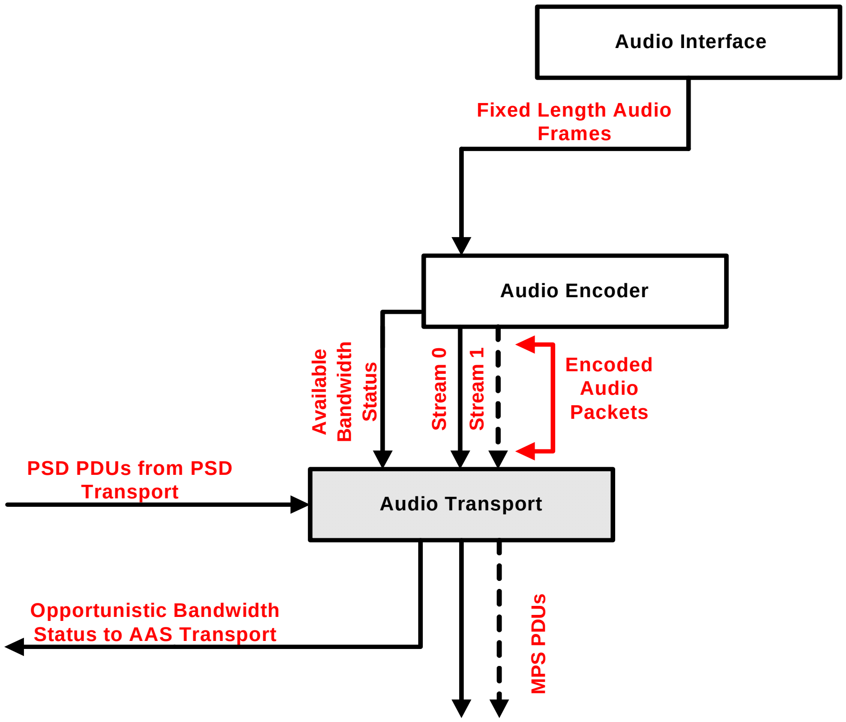


Figure 4-1: Audio Transport Interface Diagram

4.2 Audio Encoder

The Audio Encoder is a block-processing algorithm; each block or audio frame corresponds to 2048 input samples from each channel (for example, left and right), regardless of the number of channels. However, the audio codec may be a variable bit rate process. In this case, each input audio frame corresponds to a variable length output encoded audio packet. Therefore, the codec must also employ a built-in rate-control mechanism so that these packets can easily be transmitted over a constant capacity channel. The rate-control mechanism works in conjunction with suitable buffering within the audio codec. The audio codec rate-control and buffer-control mechanisms ensure that the implemented buffering is sufficient (that is, the buffers will not overflow or underflow). With these mechanisms, the audio codec may be treated as a fixed rate codec with a specified constant delay between the input and the output.

Although the encoded audio packet size variability has an impact on the system in terms of variable tuning delay (smaller packets can be decoded faster than longer packets), there is almost no additional cost for the system in terms of increased system complexity. The packet size variability is essentially invisible to the other parts of the system and manifests itself only in terms of an additional constant audio codec delay.

The Audio Transport supports a multi-stream codec. The HD Radio system may utilize multi-stream audio transmission to provide robust coverage and fast tuning times. A multi-stream Encoder segregates the encoded audio content into separate bit-streams. The "more important" encoded bits are placed in a *core* bit-stream such that it is independently decodable, albeit at reduced audio quality. The remaining bits are placed in an *enhanced* bit-stream which will, when combined with the core bit-stream at the decoder, produce an audio output at a level of quality that is substantially identical to that of a decoded single stream at a bit rate equivalent to the total bit rate of the core and enhanced bit-streams. The enhanced bit-stream is not independently decodable.

Audio frames are received from the Audio Interface and are processed by the Audio Encoder. Based on the configuration, encoded audio packets are written to a separate memory buffer for each Audio Encoder output stream. Figure 4-2 shows the Audio Encoder Interfaces. The encoded audio packets are sent to the Audio Transport.

The peak audio level shall be matched to the implemented audio codec to prevent clipping and other distortion. The exact audio level is dependent on the codec specifications.

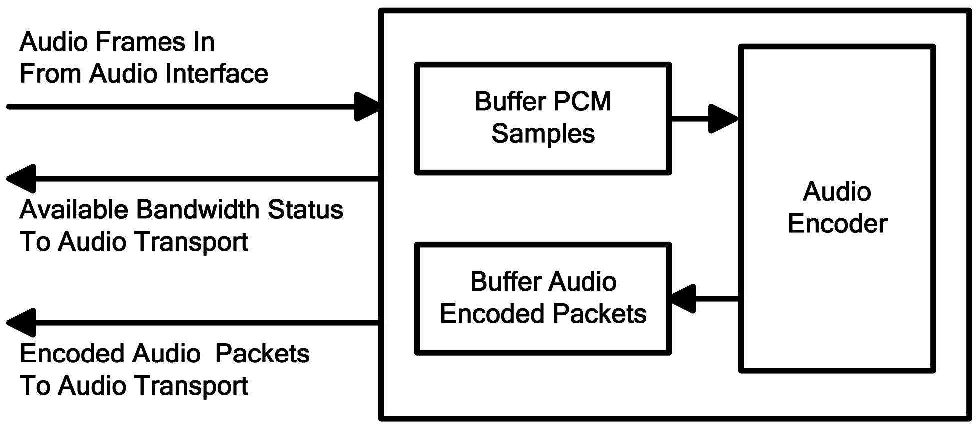


Figure 4-2: Transmit Audio Encoder Interface

4.3 Audio Transport

The Audio Transport accepts the variable length compressed audio packets from the Audio Encoder and packs them into fixed length PDUs. On average, the number of encoded audio packets per PDU is N. Refer to Table 5-2 for the value of N for different audio codec modes. However, this may vary by an elastic-buffer size parameter, *D,* from one PDU to another to account for the packet size variability. Denoting the number of encoded audio packets per PDU as n, the number of packet per PDU will be with in the range (N-D) ≤ n ≤ (N+D). For example, if N=32 and D = 8, each PDU may contain from 24 to 40 encoded audio packets, with an average of 32. Figure 4-3 illustrates this example. It must be noted that the PDU bit length has a maximum size based on the L1 rate and the codec rate. However, the number of audio packets is variable to accommodate extra throughput per packet. The parameter D is sent over-the-air as part of the MPS PDU and is used to determine the amount of buffering needed at the Audio Decoder to make the encode/decode process appear as a fixed delay process. Sequence numbers are also sent over the air so that the audio decoder can perform proper alignment between multiple streams.

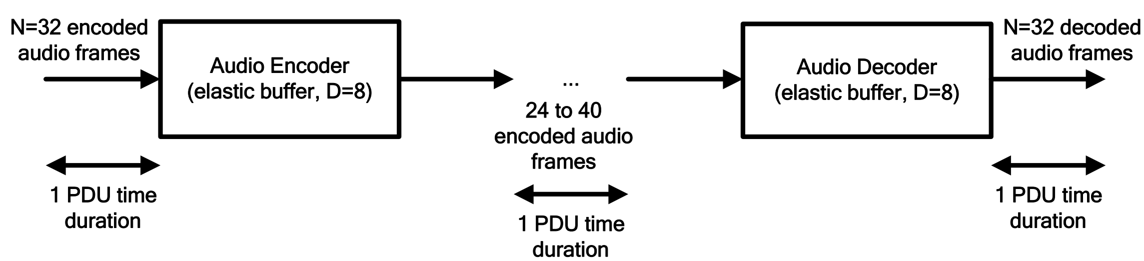


Figure 4-3: Elastic Buffering Example

MPS PDUs are always produced in time units of 4 or 32 input audio frames. In some audio codec modes, multiple encoded audio streams are produced. The MPS PDU rate may be different for each of the streams; for example, 4 audio frame times for the core stream and 32 audio frame times for the enhancement stream. In this example, a valid enhancement PDU will be generated only once for every eight valid core PDUs.

In generating an MPS PDU, the compressed bits corresponding to each audio frame are buffered until an encoded output unit is complete (after 4 or 32 input audio frame times) and an MPS PDU is constructed with a maximum length determined by the configuration, as shown in Figure 4-4.

At the receiver, the encoded packets for each stream (core and enhanced streams must be aligned), are input to the Decoder. The Audio Decoder then decodes and outputs the N audio frames contained within the encoded packets. This assumes that the rate control mechanism for the codec variability is implemented within the Decoder. In addition, appropriate buffering of encoded packets is implemented by the Audio Transport such that a constant delay is maintained through the encode/decode process.

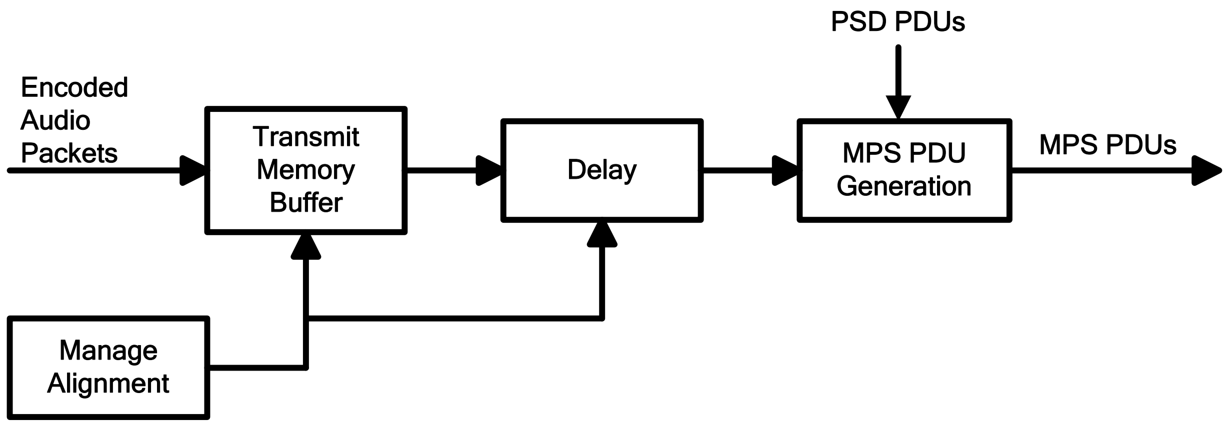


Figure 4-4: Audio Transport Block Diagram

4.3.1 Blending

The blend mechanism in the Hybrid AM and Hybrid FM systems accurately aligns the analog signal and the digital signal in time. The interface shown in Figure 4-5 allows the Audio Decoder in the receiver to be treated as a constant delay element. In other words, the time between providing MPS PDU data to the Decoder and corresponding output audio generation (that is, the audio for which the Encoder returned this MPS PDU) is a constant D (exclusive of Decoder implementation delay, Δ), as selected by the encoder. Typical Decoder delays include processing delay and error mitigation delay.

The delay is constant irrespective of the start (tune-in) time at the Decoder or the characteristics of the audio system. It is also constant regardless of the nature of the audio or the starting MPS PDU at the decoder.

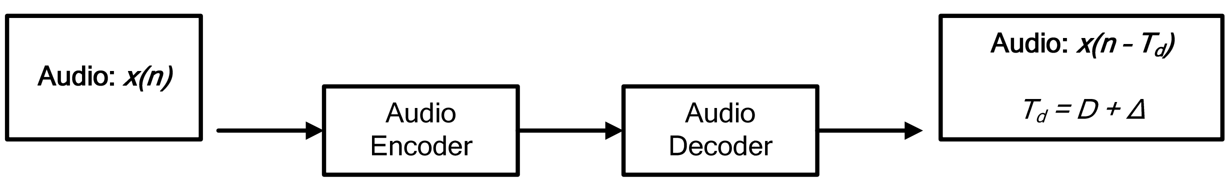


Figure 4-5: Audio Encoder/Decoder as a Constant Delay Element for the System

4.3.2 Additional Delay Compensation

The Audio Transport interface must support a provision for additional delay compensation both at the transmitter and the receiver. This is in addition to the automatic delay compensation inherent in the Audio Decoder related to the frame size variability. At the transmitter, this compensation may be requested in units of PDUs, in which case the encoded audio stream is delayed by the corresponding number of PDUs prior to being sent.

4.4 Data Transport Interface

The Audio Transport provides a mechanism for (byte-oriented) opportunistic and PSD capability. The Audio Transport mechanism consists of: the following:

- Interface for injecting PSD into an MPS PDU at the Encoder.
- Notification mechanism for Encoder-side entity (when more data is needed).
- Notification mechanism for Encoder-side entity of unused capacity.

4.4.1 Program Service Data

The audio codec PSD facility is a variable byte-stream and exists, though is not necessarily used, for each defined audio stream. A PSD byte-stream may be added to each individual MPS PDU in time-varying amounts. This is enabled through the PSD Transport [9]. Refer to Subsection 5.2.2 for details on PSD processing.

4.4.2 Opportunistic Data

The audio encoder provides a mechanism to indicate any unused byte-oriented capacity within the MPS PDU or the SPS PDU which is then made available for other data applications in the HD Radio system. The unused byte capacity as determined by the audio encoder is available for use by the AAS Transport and the transmission of opportunistic data services. The unused capacity is indicated per encoded stream. The unused bandwidths from MPS and SPS in each logical channel are aggregated for this service. This bandwidth is allocated on a PDU-per-PDU basis and may not be available in every consecutive PDU. The data rates for Audio Encoder opportunistic data streams are heavily dependent on the audio program, and may vary from zero to several kbit/s.

The Available Bandwidth Status indication is dependent on the actual system software implementation. The HD Radio system developed by iBiquity provides the size of each encoded packet in each stream to other processes in the exciter.

5 Protocol Data Unit

5.1 Protocol Data Unit Configuration

In order to achieve efficiency with respect to system throughput and data bandwidth, it is assumed that the Audio Transport derives certain configuration information through administrative primitives. Such configuration information is derived from other parts of the system, such as the Audio Interface at the transmitter or in the process of relaying MPS PDUs to the Audio Transport at the receiver. This configuration is affected by the primary service mode of the system. All bit rates, as indicated in tables and figures in this section, are approximate and usually rounded to the nearest thousand.

Once the desired configuration information is available, the Audio Transport (at the transmitter) accepts audio samples at the Audio Interface and converts them into an MPS PDU.

5.2 PDU Characteristics

The exact handling of the audio frames and the resulting PDU is uniquely defined by a combination of audio codec mode and stream number. That combination is included in the PDU information for retrieval and proper handling by the audio decoder.

5.2.1 PDU Structure

Figure 5-1 illustrates the PDU format for all audio codec modes. This PDU format includes a fixed header portion (Control Word), a variable number of Audio packet location fields, an optional Expansion field, PSD, and encoded audio packets. The Control Word is protected by a 96 byte RS code. Since the RS codeword is a fixed size it may also span portions of the Expansion field, PSD field, and possibly the encoded audio packets.

Each MPS PDU consists of RS parity bytes, the MPS PDU Control Word, the Audio packet locators, optional header expansion, and a variable number of encoded audio packets. Each locator points to the CRC byte following the packet it covers, using one locator per packet. The size of the locator field (Lc bits) is a variable and is optimized (matched) to the PDU length to reduce overhead. Each encoded audio packet is protected by an 8-bit CRC field.

The internal buffer control mechanism of the Audio Encoder handles the fractional packet scenario (where the first or last packet generally spans across two MPS PDUs). This scenario is necessary to ensure that the MPS PDU is of a certain maximum size. The Audio Transport ensures that this is transparent to other layers. The first byte of the PDU is defined as byte 0; the last byte of the PDU depends on the variable size of the PDU.

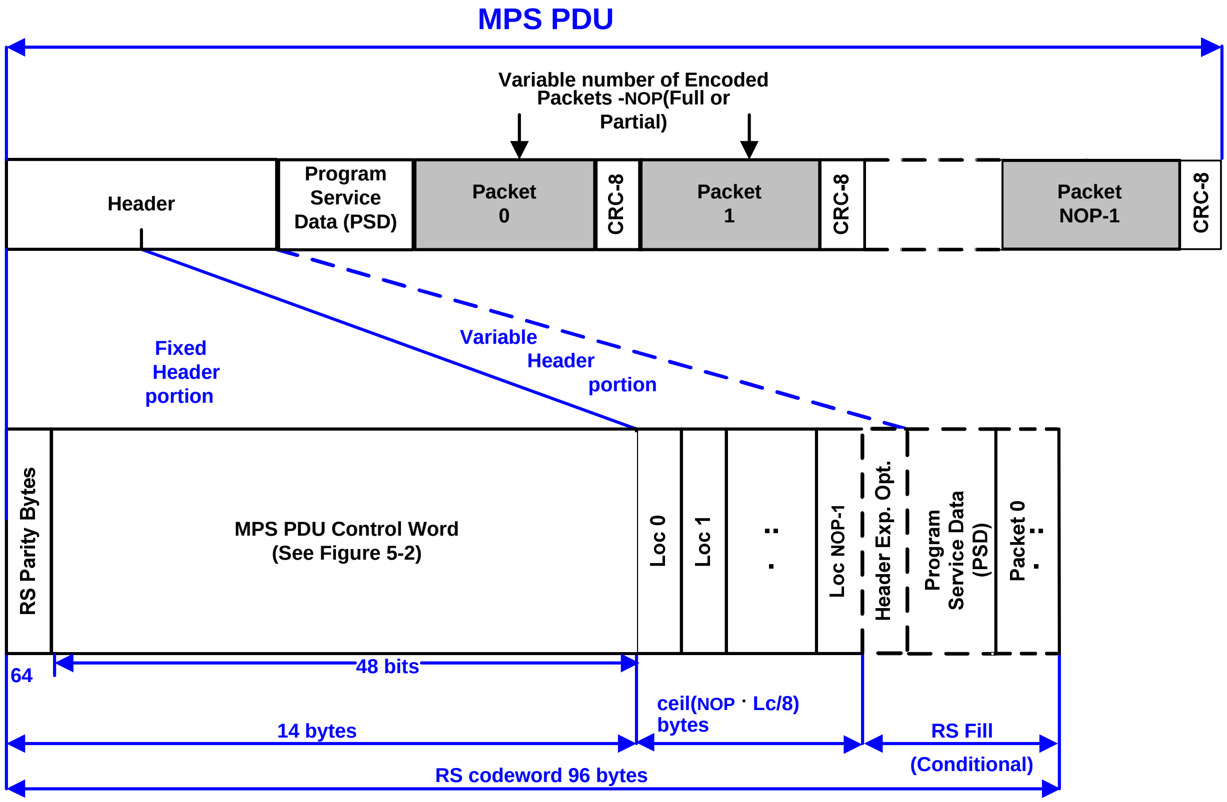


Figure 5-1: MPS PDU Format

The fields enclosed by the dotted lines (Header Expansion, PSD, portion of encoded audio packets) are part of the RS codeword only conditionally. The overall length of the RS parity, control word, Loc indicators may be less than the 96 byte RS codeword. When this case occurs, additional fields (Expansion header, PSD, Packet 0) fall within the 96-byte codeword and will be included in the RS parity byte computation.

Figure 5-2 shows the MPS PDU control word. It shows the bit allocation within the control word and the bit aggregation into bytes.

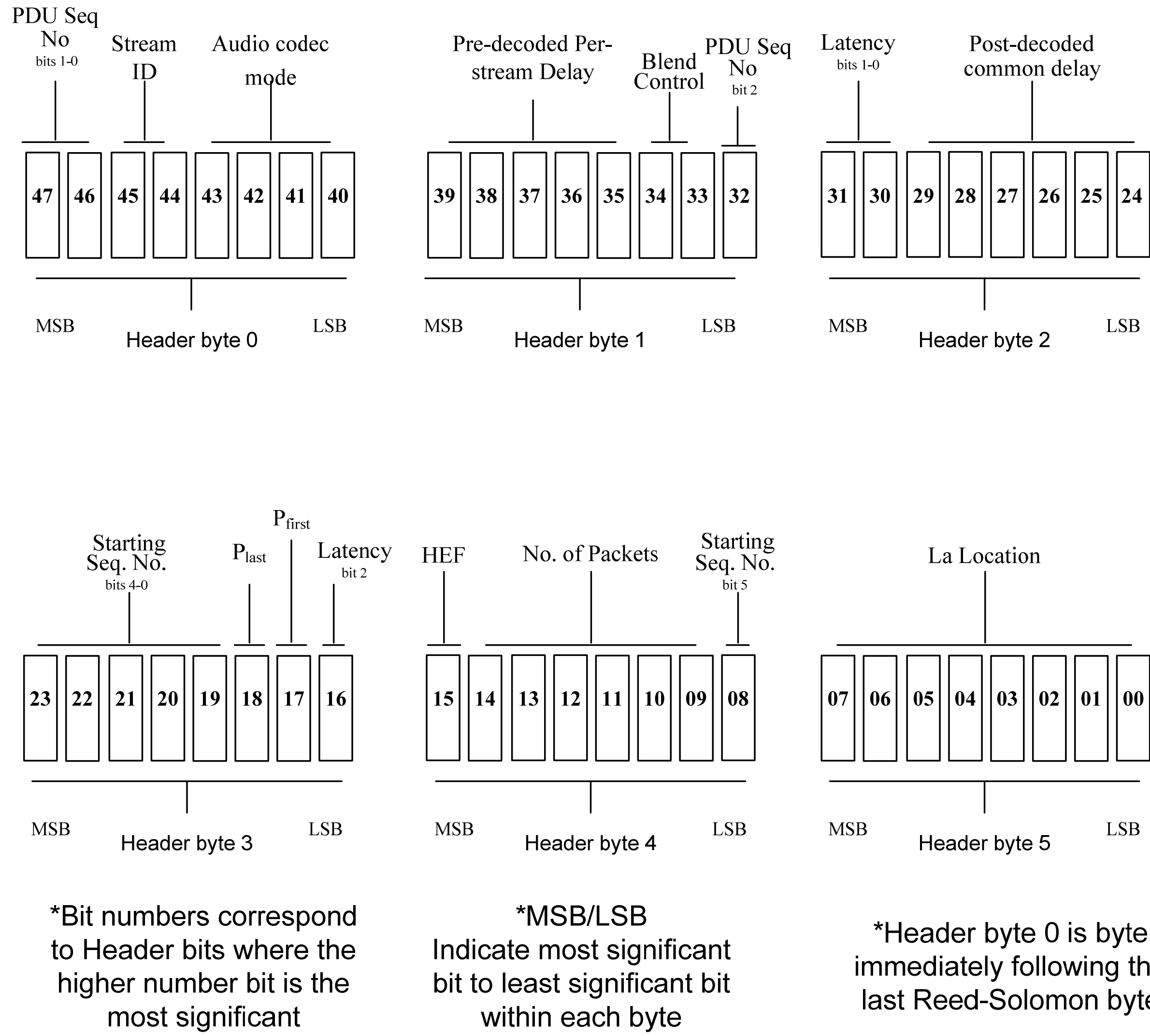


Figure 5-2: MPS PDU Control Word – Bit Allocation

Details of the PDU format are presented in Table 5-1 and the ensuing subsections.

Table 5-1: MPS PDU Header Field Definitions

Bits	Description	Comments
64	Parity Bits	Eight RS parity bytes for header error protection
4	Audio Codec Mode	See Table 5-2. This parameter must be set to the same value for all streams.
2	Stream ID	This field is used to indicate the stream type Core (00) Enhanced (01) Reserved (10) and (11)
3	PDU Sequence Number	This is the sequence number of the PDU It increments by one modulo the sequence number range (as defined in Table 5-2) every PDU
2	Blend Control	This field must be set to the same value for all streams. Refer to Table 5-5.

Bits	Description	Comments
5	Pre-decoded Per Stream Delay/ TX Digital Audio Gain	If Stream ID = 00, this field defines the TX Digital Audio Gain (see Table 5-6). For all other Stream IDs, this field defines the delay between this corresponding stream and stream 00. Stream 00 is never delayed but the corresponding stream which is delayed. This delay, in units of 4 audio frame periods, is that which the receiver is to normally apply prior to audio decoding.
6	Post-decoded Common Delay	Delay, in units of 4 audio frame periods, that the receiver uses to align the digital audio content with associated analog audio content after audio decoding. This field must be set to the same value for all streams.
3	Latency	Audio codec latency in units of 2 audio frame periods; the value is limited to a maximum value of 10. This value constrains the total audio encoder/decoder delay to a fixed value. This parameter excludes any decoder implementation delays and is equivalent to the elastic buffer depth, D. This field must be set to the same value for all streams.
1	P_{first} Flag	First Packet Partial P_{first} is set to “1” if the first encoded audio packet of the PDU is a continuation from the last PDU.
1	P_{last} Flag	Last Packet Partial P_{last} is set to “1” if the last packet of the PDU will continue into the next PDU.
6	Starting Sequence Number	Sequence number of the encoded audio packet in the PDU following the first partial packet, if any. Initialized to zero for all streams simultaneously; the range is 0 to 63.
6	Number of Packets (NOP)	Number of encoded audio packets contained within the PDU, full and partial.
1	Header Expansion Flag (HEF)	Header Expansion Flag is set to “1” when optional Header Expansion Field is inserted immediately following Location Fields.
8	La Location	Location of last byte of the Program Service Data field. The location is relative to the first byte (0) of the PDU.
NOP·Lc	Locator fields	Each of NOP pointers consisting of Lc bits points to location of last byte (CRC location) of the corresponding packet for each of the NOP packets (partial or full) in the PDU. The location is relative to the first byte (0) of PDU.
0 - 128 (optional)	Header Expansion Field(s) (optional)	Header Expansion Field present only when Header Expansion Flag is set to “1”. See Table 5-7 for format.
RSfill (Conditional)	Start at byte 14 + ceil(NOP·Lc/8) + (Number of Header Expansion Fields) through byte 95	This field extends to the end of the 96-byte RS codeword. Bytes include the start of Expansion bytes, PSD field, followed by Audio packets when the Expansion bytes, PSD do extend to the RS codeword length (96-bytes).

5.2.1.1 Audio Codec Mode

Table 5-2 defines the FM and AM audio codec modes according to stream configuration, frames, and bit-rates.

Table 5-2: Audio Codec Mode Definitions

Audio Codec Mode	Typical Use	Number Of Streams	Stream ID	Stream Type Core or Enhanced)	PDUs Per L1 Frame	Average Number of Encoded Audio Packets Per PDU (N)	PDU Sequence Number Range	Lc bits Per Location	Nominal Bit Rate (kbit/s)
0000	FM Hybrid	1	00	Core	1	32	0 - 1	16	96
0001	FM All Digital	2	00	Core	8	4	0 - 7	12	48
			01	Enhanced	1	32	0 - 1	16	48
0010	AM Hybrid	2	00	Core	8	4	0 - 7	12	20
			01	Enhanced	1	32	0 - 1	16	16
	AM All Digital	2	00	Core	8	4	0 - 7	12	20
			01	Enhanced	1	32	0 - 1	16	20
0011	FM All Digital	2	00	Core	8	4	0 - 7	12	24
			01	Enhanced	1	32	0 - 1	16	72
1010	FM	2	00	Core	1	32	0 - 1	12	22
			01	Enhanced	8	4	0 - 7	12	24
1101	FM	1	00	Core	8	4	0 - 7	12	24
0100 - 1001 1011 - 1100 1110 - 1111	Reserved	-	-	-	-	-	-		-

The audio codec output bit-rate can be scaled to provide additional capacity for other applications. The audio codec throughput is limited by the maximum PDU lengths for the different service modes as specified in [1] and [2]. Table 5-3 shows the minimal bit-rate allowed for each audio codec mode.

Table 5-3: Defined Output Bit Rates and Streams for the Audio Codec

Audio Codec Mode	Nominal Bit Rate (kbit/s)	Minimum Bit Rate (kbit/s)[1]
0000	96	24
0001	96	24
0010	36 / 40[2]	20
0011	96	24
1010	46	22
1101	24	12

[1] These values reflect useful rates. "0" rate could also be considered, but that reflects an inactive codec. Also future configurations may result in different values.

[2] Depends on the actual system configuration

Table 5-2 shows there exists audio codec modes that are not yet defined. However, all future audio codec modes must maintain backward compatibility with certain streams of the defined modes. Table 5-4 shows the stream compatibility for all the reserved codec modes.

Table 5-4: Reserved Audio Codec Modes – Defaults

Audio Codec Mode	Default Audio Mode	Backward Compatible Streams	Stream Free to be Redefined[†]
0100	0010	00,01	-
0101	0010	00,01	-
0110	0010	00	01
0111	0010	00	01
1000	0000	00	-
1001	0011	00,01	-
1011	0011	00	01
1100	0001	00,01	-
1110	0001	00	01
1111	-	-	-

[†] Additional streams are assumed in expanded all-digital system service modes and possibly in advanced (future) hybrid configuration.

Table 5-2 defines the audio codec modes according to stream configuration, frames, and bit rates. Single-stream audio codec modes, while at the nominal rate, contain at least 2-channel audio (stereo). This also applies to multiple-stream systems when all streams are valid at the decoder input. However, if just the primary stream of a multiple-stream is valid at the decoder input, the audio may be either stereo or mono depending on the audio codec mode. A monophonic signal (L&R) is always decoded or presented as a dual-mono (that is, identical left and right channels) signal by the audio decoder.

5.2.1.2 Blend Control

Table 5-5 defines the blend control bits. This definition only applies to the Main Program Service Audio (MPSA); for any Supplemental Program Service Audio (SPSA) the bits are set to “00”.

Table 5-5: Blend Control Bit Definitions

Audio Control Word Bit 34	Audio Control Word Bit 33	Waveform	Service Mode	Definition
0	0	AM & FM Hybrid	MP1 MP2 MP3 MP11 MA1	Not Valid
		AM & FM All Digital	MP5 MP6 MA3	No analog diversity delay has been applied by the transmitter. RX shall disable analog blending. This should always be sent by the broadcaster when in any all digital service mode.
0	1	AM & FM Hybrid FM Extended Hybrid	MP1 MP2 MP3 MP11 MA1 MP5 MP6	No analog diversity delay has been applied by the transmitter. RX shall disable analog blending.
		AM & FM All Digital	MP5 MP6 MA3	Not Valid
1	0	AM & FM Hybrid	MP1 MP2 MP3 MP11 MA1	Analog diversity delay has been applied by the transmitter. RX shall blend to analog when the digital audio quality measure is below the selected threshold.
		AM & FM All Digital	MP5 MP6 MA3	Not Valid
1	1	All	MP1 MP2 MP3 MP11 MP5 MP6 MA1 MA3	Reserved

5.2.1.3 Pre-decoded Per Stream Delay / TX Digital Audio Gain

Table 5-6 defines the bit values of the "Pre-decoded Per Stream Delay / TX Digital Audio Gain" when this field is used to define the offset to be applied to the digital audio level at the receiver to equalize the subjective loudness of the digital and analog audio.

Table 5-6: TX Digital Audio Gain Control

Value	RX Digital/Analog Audio Adjustment
Reserved	…
11000	-8 dB
11001	-7 dB
11010	-6 dB
11011	-5 dB
11100	-4 dB
11101	-3 dB
11110	-2 dB
11111	-1 dB
00000	0 dB
00001	+1 dB
00010	+2 dB
00011	+3 dB
00100	+4 dB
00101	+5 dB
00110	+6 dB
00111	+7 dB
Reserved	…

5.2.1.4 Header Expansion Flag

The Header Expansion Flag (HEF) is used to indicate the presence of additional fields within the header. The HEF is set to "1" when there is a Header Expansion field present. The first Expansion field is inserted immediately following the Locator Fields. Refer to Subsection 5.2.1.6 for a detailed description of the optional Header Expansion Fields.

5.2.1.5 Locator Fields

The bit organization of 16-bit and 12-bit locator fields is shown in Figure 5-3 and Figure 5-4 respectively. Each of the NOP locators points to the location of the last byte of the corresponding packet for each of the NOP packets (partial or full) in the PDU. The location is relative to the first byte of PDU (0). Each 16-bit locator consists of 2 bytes. The 12-bit locators consist of one byte and a nibble. The next 12-bit locator begins with the next nibble. Bit b_{NOP-1}^{0} is sent first and bit b_{NOP-1}^{15} is sent last. The bit length of the locators is defined by the audio codec mode as shown in Table 5-2.

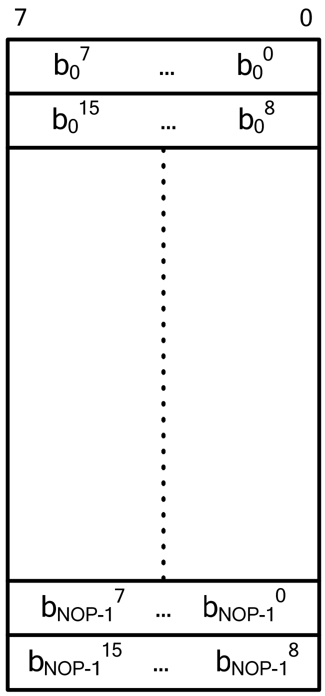
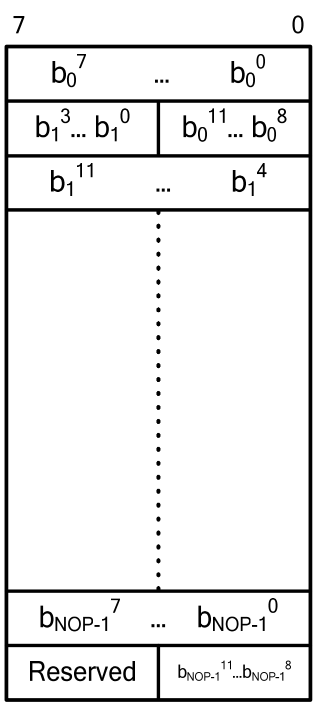


Figure 5-3: Locator Fields – 16-Bit



* is unused when NOP is odd

Figure 5-4: Locator Fields – 12-Bit

5.2.1.6 Header Expansion Fields

The header expansion field bits are as defined in Table 5-7.

Table 5-7: Header Expansion Field Bits – Definitions

<table>
<tr><th>Bits</th><th>Description</th><th colspan="2">Comments</th></tr>
<tr><td>1</td><td>HEF
Header Expansion Flag</td><td colspan="2">Header Expansion Flag is set to “1” when optional additional Header Expansion Fields are inserted following immediately.</td></tr>
<tr><td rowspan="9">3</td><td rowspan="9">Header Expansion ID</td><td>Bits</td><td>Header Expansion Format Type</td></tr>
<tr><td>000</td><td>Reserved</td></tr>
<tr><td>001</td><td>Program Number indication
This value for the Header Expansion ID field indicates that the Program Number is provided according to the description in Subsection 5.2.1.6.1.1</td></tr>
<tr><td>010</td><td>Program Type indication
This value for the Header Expansion ID field indicates that the Program Type is provided according to the description in Subsection 5.2.1.6.2.1</td></tr>
<tr><td>011</td><td>Reserved.</td></tr>
<tr><td>100</td><td>Reserved</td></tr>
<tr><td>101</td><td>Reserved</td></tr>
<tr><td>110</td><td>Reserved</td></tr>
<tr><td>111</td><td>Reserved</td></tr>
<tr><td>3</td><td>Expansion Content</td><td colspan="2">Header Expansion Content – depends on the Header Expansion ID.</td></tr>
<tr><td>1</td><td>LSB</td><td colspan="2">This bit is defined based on the Header Expansion ID.</td></tr>
</table>

As shown in Table 5-7, the Header Expansion Field also contains a one bit HEF. This flag serves the same purpose as the HEF in the main header, that is, to indicate the presence of additional header fields. It is set to “1” when an additional Header Expansion Field is present. All additional Expansion fields are inserted consecutively after the first Expansion field. Each Header Expansion field consists of 1 byte. Each HEF indication is associated with the next header expansion byte immediately following. The maximum number of header expansion fields is limited to 16. In streams containing PSD, the expansion bytes are transmitted instead of the PSD. If there is no PSD present or if the expansion bytes exceed the allocated PSD capacity, the expansion bytes are transmitted instead of audio packets. The location of the last byte of PSD, La, points to the last location of the Expansion Fields when it exceeds the amount of PSD.

Figure 5-5 shows the Header Expansion Flag and the indication of additional header expansion fields.

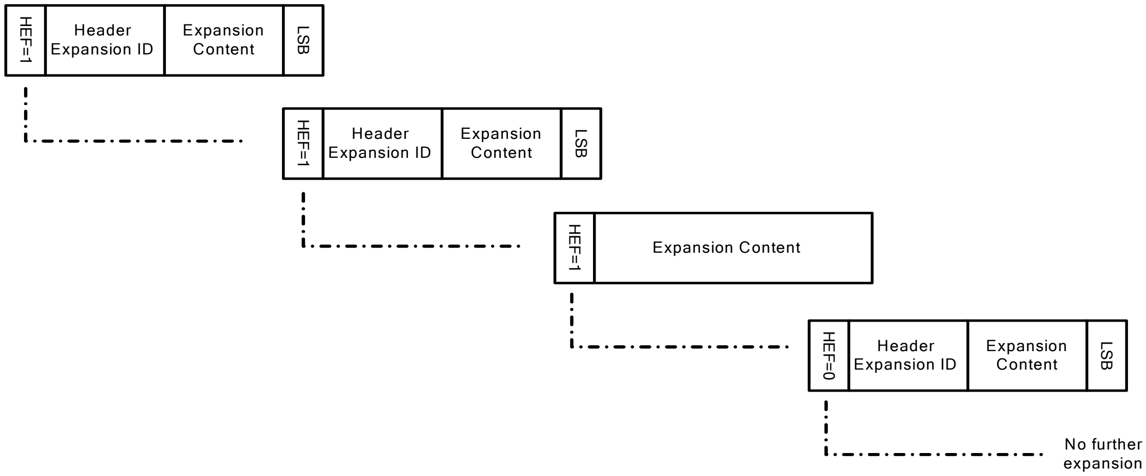


Figure 5-5: Header Expansion Fields – Example

Both the MPS and the SPS are characterized by the indications – Program Number, Program Type.

The Header Expansion ID is used to show what type of indication is provided. The Expansion Content field is primarily used to define the type of indication provided. As shown in Table 5-7, the format of the Expansion Content field depends on the Header Expansion ID and may contain the following indications:

a) Program Number – To identify and manage programs

b) Program Type – To identify the different program types

The Program Number can be a value between 0 and 7; however Program Number “0” always designates the MPS. Refer to Subsection 5.2.1.6.1 for a detailed description of the Program Number indication. The Program Type can be of any type for all the programs.

Table 5-8: Program Indications for MPS/SPS

Indication	MPS	SPS
Program Number	0	1 - 7
Program Type	Any type (Table 5-9)	Any type (Table 5-9)

MPS needs to include the Program Number if additional indications such as Program Type and Program ID are conveyed whereas SPS does need to indicate the Program Number at the minimum.

A program does not necessarily have to include these indications and a “non-present” indication would be considered as “zero”.

Just as the Expansion Content, the LSB of the Header Expansion Field is also defined based on the Header Expansion ID and the type of indication – Program Number, Program Type.

5.2.1.6.1 Program Number

If the Header Expansion ID is set to “001” (Program Number indication), then the Expansion Content contains the Program Number for the current stream.

The Program Number indication is used by the Audio Transport to identify and manage the MPS or SPS programs transmitted as shown in Table 5-8. Program Number “0” is used only for the Main Program Service. SPS can be designated using any Program Number (1 through 7) by the broadcasting system and transmitted in any order as required. A supplemental program can be added or removed without affecting the main program or existing supplemental programs being transmitted. The Program Number is always indicated first using the Header Expansion fields before the Program Type or Program ID. For MPS, the Program Number is indicated only if additional Header Expansion fields are used to indicate Program Type or Program ID. In this case, it is indicated first using the Header Expansion fields before the Program Type or Program ID. For SPS, the Program Number is always indicated and is the first indication using the Header Expansion fields before the Program Type or Program ID.

5.2.1.6.1.1 Header Expansion for Program Number

The Header Expansion for the Program Number is shown in Figure 5-6.

HEF	001	Program number 0...7	Reserved

Figure 5-6: Header Expansion – Program Number

The Expansion Content contains the Program Number (0 - 7).

5.2.1.6.2 Program Type

The Program Type is indicated in the Expansion Content when the Header Expansion ID is set to "010". The Program Type indication represents the program content. It enables the receiver to search and sort through the variety of program content being broadcast. The program types defined for use in the HD Radio system are shown in Table 5-9. This table represents the most commonly used program types for broadcast and fully complies with the RBDS definition. Additional Program Types can be included as required. The Program Types shown in Table 5-9 are independent of the audio codec and its characteristics and shown only for reference.

Table 5-9: Audio Program Type Indications

Type number	Program type
0	Non specific / Undefined
1	News
2	Information
3	Sports
4	Talk
5	Rock
6	Classic Rock
7	Adult Hits
8	Soft Rock
9	Top 40
10	Country
11	Oldies
12	Soft
13	Nostalgia
14	Jazz
15	Classical

Type number	Program type
16	Rhythm and Blues
17	Soft Rhythm and Blues
18	Foreign Language
19	Religious Music
20	Religious Talk
21	Personality
22	Public
23	College
24 - 28	Unassigned / Future use
29	Weather
30	Emergency Test
31	Emergency
32 - 63	Unassigned / Future use
64	Weather B
65	Traffic A
66	Traffic B
67	Traffic C
68 - 72	Unassigned / Future use
73	Government
74	Emergency B
75	Emergency C
76	Special Reading Services
77 - 82	Unassigned / Future use
83	Private Services A
84	Private Services B
85	Private Services C
86 - 255	Unassigned / Future use

For a multi-stream configuration, the Program Type is indicated only on the core (main) stream.

Different programs are assigned the Program Type by the broadcasting system and it is optional. The Program Type information broadcast can be used in a variety of ways from initial screening to prioritizing and sorting out programs. The Program Type is transmitted at least once per modem frame and is required only on the core stream in multi-stream configurations.

Concurrently, the same Program Type indication is optionally included by the broadcast system in the fast-acquired SIS information. This is achieved by adding additional fields to the SIS PDUs passed through the PIDS logical channel. Refer to [7] for details. Here, it can be used for faster searching/scanning for available and/or desired programs.

The Program Type header expansion can be changed without any interruption to the current program.

5.2.1.6.2.1 Header Expansion for Program Type

The header expansion for Program Type is shown in Figure 5-7.

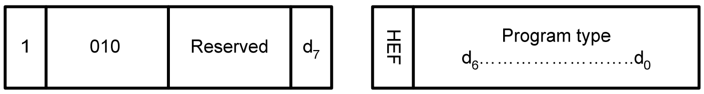


Figure 5-7: Header Expansion – Program Type

The Program Type consists of two expansion bytes. The first byte contains the Header Expansion ID for the Program Type. The Program Type number is then constructed in the next expansion byte which contains the 7 LSBs of this Program Type number. The LSB of the first byte is then used as the MSB to construct the actual 8-bit Program Type number. The 3-bit Expansion Content of the first byte is Reserved and may be used to further expand the Program Types.

5.2.2 Program Service Data (PSD) Processing

PSD is available within the Audio Transport layer. The Audio Transport interface accepts PSD PDUs from the PSD Transport [9]. A PSD byte-stream is supported by each active encoder stream but utilized for the core stream only. Refer to Table 5-10 for the minimum guaranteed and typical/average number of data bits per PDU per stream for each of the various audio codec modes.

Table 5-10: Program Service Data per Audio Codec Mode

Audio Codec Mode	Typical Use	Stream ID (bits)	Stream Type	Nominal[2] Program Service Data Rate (Bytes/PDU)
0000	FM Hybrid	00	Core	128
0001	FM All Digital	00	Core	7
		01	Enhanced	0
0010	AM Hybrid / All Digital	00	Core	7
		10	Enhanced	0
0011	FM All Digital	00	Core	7
		01	Enhanced	0
1010	FM	00	Core	7
		10	Enhanced	0
1101	FM	00	Core	7
0100 - 1001 1011 - 1100 1110 - 1111	Reserved	-	-	-

[2] Rates are not guaranteed and may depend on the audio format and the actual bit rate.

In AM, the supplemental programs (SPS) do not carry Program Service Data.

5.2.3 Error Control Codes

Error control codes are utilized in various portions of the PDU in order to provide for error detection and/or correction in the receiver. The codes included are: Reed Solomon (RS) and cyclic redundancy check (CRC).

5.2.3.1 Packet Locator Protection

In all streams within all audio codec modes, the header payload is protected by an error correction (and detection) code. The code in use is RS of GF (2^8). The actual code word is shortened to a length of 96 bytes - (96, 88, 2^8). Each codeword consists of the header payload bytes along with 8 redundancy (parity) bytes. The header payload is described in Figure 5-1.

- Primitive polynomial is $x^8+x^4+x^3+x^2+1$ (100011101 in binary notation, most significant bit on the left)
- Generator polynomial is

  $$g\ (x) = a^{36} + a^{203}x + a^3x^2 + a^{220}x^3 + a^{253}x^4 + a^{211}x^5 + a^{240}x^6 + a^{176}x^7 + x^8$$

  where "*a*" is a root of the primitive polynomial.
- To compute the parity bytes, it is assumed that bytes 0 through 158 of the un-shortened input codeword are zero. Byte 160 is the rightmost byte shown in Figure 5-1. Byte 247 of the RS codeword is the first byte (leftmost) of the MPS PDU Control Word shown in Figure 5-1. The parity bytes are then computed, where the last parity byte of the RS codeword is the first byte (leftmost in Figure 5-1) in the MPS PDU.

5.2.3.2 Packet Integrity Control

Each encoded audio packet is accompanied by a CRC-8 code for the purpose of receiver integrity check. The generator polynomial used is:

$$g_8(x) = x^8 + x^5 + x^4 + 1$$

This polynomial can be represented in binary form as 100110001 where the LSB is on the right. The CRC value is computed as follows:

> Perform modulo-two division of the encoded audio packet by the generator polynomial $g_8(x)$. The 8-bit remainder inserted into the PDU will have the least significant bit directly following the last bit of the data packet.

GLOSSARY

Audio Frame
The unit of information payload exchanged from the Audio Interface and the Audio Transport Layer.

Audio frames are comprised of 2048 audio samples at a sampling rate of 44.1 kHz.

Audio Quality
High audio quality is required by the system specification for each of the primary L1 service modes while maintaining the necessary compression rate.

Audio Encoder
Audio Encoder refers to the audio processing at the transmission side only.

On the other hand, audio codec refers to the combined transmit and receive audio processing functions in the system.

MPS PDU
Refers to the output of the Audio Transport process.

An MPS PDU consists of protocol information followed by a sequence of encoded audio packets.

MPS PDUs may be output from one to two streams depending on the audio codec mode.

Encoded Audio Packet
Compressed audio frames output from the Audio Encoder.

These may be divided into one to two output streams depending on the audio codec mode.

Layer 1 (L1)
The lowest protocol layer in the HD Radio Protocol Stack

Also known as the waveform/transmission layer

Primarily concerned with the transmission of data over a communication channel.

Includes framing, channel coding, interleaving, modulation, etc. over the AM radio link at the specified service mode.

Layer 2 (L2)
The Channel Multiplex layer in the HD Radio Protocol Stack.

Multiplexes data from the higher layer services into logical channels (partitioned into L1 frames, block pairs, and blocks) for processing in Layer 1.

Main Program Service (MPS)
The Main Program Service preserves the existing analog radio-programming formats in both the analog and digital transmissions.

In addition, Main Program Service includes digital data, which directly correlates with the audio programming.

Supplemental Program Service (SPS)
Supplemental Program Service is a secondary program broadcast simultaneously with the main program using any logical channel.

Multi-stream
Audio information, split into two individual streams of encoded audio packets.

This capability is necessary to support both fast tuning and graceful degradation requirements.

Protocol Control Information (PCI)	Protocol Control Information (PCI) • Stream ID for the associated payload (that is, MPS PDU) • Length(s) of associated payload • Cyclic Redundancy Check (CRC) for the PCI
Protocol Data Unit (PDU)	A Protocol Data Unit (PDU) is the structured data block in the HD Radio system that is produced by a specific layer (or process within a layer) of the transmitter protocol stack. The PDUs of a given layer may encapsulate PDUs from the next higher layer of the stack and/or include content data and protocol-control information originating in the layer (or process) itself. The PDUs generated by each layer (or process) in the transmitter protocol stack are inputs to a corresponding layer (or process) in the receiver protocol stack.