NRSC-5-E Reference Document List

No.	Updated for Rev. E	Doc. No.	Doc. Name	Ver.	Date	Pages in .pdf Start	End	# of pages
1	X	SY_IDD_1011s	HD Radio Air Interface Design Description Layer 1 FM - NRSC	H	November 2022	2	175	174
2		SY_IDD_1012s	HD Radio Air Interface Design Description – Layer 1 AM	G	2016-12-14	176	254	79
3	X	SY_IDD_1014s	HD Radio Air Interface Design Description Layer 2 Channel Multiplex - NRSC	K	November 2022	255	285	31
4	X	SY_IDD_1017s	HD Radio Air Interface Design Description Audio Transport	I	November 2022	286	330	45
5	X	SY_IDD_1020s	HD Radio Air Interface Design Description Station Information Service Transport	K	November 2022	331	392	62
6	X	SY_SSS_1026s	HD Radio FM Transmission System Specifications - NRSC	H	December 14, 2022	393	423	31
7		SY_IDD_1028s	HD Radio Air Interface Design Description – Main Program Service Data	E	2016-12-14	424	451	28
8	X	SY_IDD_1082s	HD Radio AM Transmission System Specifications	H	November 2022	452	493	42
9		SY_IDD_1085s	HD Radio Air Interface Design Description - Program Service Data Transport	D	2016-12-14	494	527	34
10		SY_IDD_1019s	HD Radio Air Interface Design Description – Advanced Application Services Transport	H	2016-12-14	528	559	32
11	X	SY_TN_2646s	Transmission Signal Quality Metrics for FM IBOC Signals	3	November 2022	560	616	57
12	X	SY_REF_2690s	Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard	3	November 2022	617	625	9
13	X	SY_IDD_4363s	HD Radio Air Interface Design Description Low-Latency Data Service Transport	A	November 2022	626	644	19

HD Radio™ Air Interface Design Description Layer 1 FM

Version H

November 2022

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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 defines the generation of *Layer 1 (L1)* FM HD Radio signals for transmission over the air to receiving equipment. It describes how control and information are passed through the Layer 1 FM air interface to generate an HD Radio signal. It focuses on the creation of the transmitted FM HD Radio signal; specific hardware and software implementation is not described.

2 Reference Documents

STATEMENT
Each referenced document that is mentioned in this document shall be listed in the following iBiquity document:

- Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard Document Number: SY_REF_2690s

3 Abbreviations, Symbols, and Conventions

3.1 Introduction

Section 3 presents the following items that are pertinent to a better understanding of this document:

- Abbreviations and Acronyms
- Presentation Conventions
- Mathematical Symbols
- FM System Parameters

Note: *A glossary defining the technical terms used herein is provided at the end of this document.*

3.2 Abbreviations and Acronyms

ALFN	Absolute L1 Frame Number
AM	Amplitude Modulation
ASF	Secondary Amplitude Scale Factor Select
ASM	Advanced Service Mode Control
BC	L1 Block Count
BPSK	Binary Phase Shift Keying
DSB1	Advanced Dual-Sideband Primary Service Mode DSB1
DSB1OV	Advanced Dual-Sideband Overlay Primary Service Mode DSB1OV
EAS	Emergency Alert System
FM	Frequency Modulation
GPS	Global Positioning System
IBOC	In-Band On-Channel
IP	Interleaving Process
kbit/s	kilobits per second
L1	Layer 1
L2	Layer 2
MF	Medium Frequency
MHz	Megahertz
MP1 – MP3, MP11, MP5, and MP6	Primary Service Modes 1 through 3, 11, 5, and 6
MP1X	Advanced Extended Primary Service Mode MP1X
MP1XOV	Advanced Extended Overlay Primary Service Mode MP1XOV
MP6OV	Advanced Overlay Primary Service Mode MP6OV
MS5	Secondary Service Mode 5
N/A	Not Applicable
OFDM	Orthogonal Frequency Division Multiplexing
P1 – P4	Primary Logical Channels 1 through 4
P3ISI	P3 Interleaver Select Indicator
PDU	Protocol Data Unit
PIDS	Primary IBOC Data Service Logical Channel
PIDSOV	Primary IBOC Data Service Overlay Logical Channel
PM	Primary Main
POV	Primary Overlay Logical Channel
PSM	Primary Service Mode Control
PSMI	Primary Service Mode Indicator
PX	Primary Extended
QAM	Quadrature Amplitude Modulation

QPSK	Quadrature Phase Shift Keying
RF	Radio Frequency
RSID	Reference Subcarrier Identification
S1	Secondary Logical Channel 1
SB	Secondary Broadband
SCA	Subsidiary Communications Authorization
SCCH	System Control Channel
SCI	Secondary Channel Indicator
SIDS	Secondary IBOC Data Service Logical Channel
SIS	Station Information Service
SM	Secondary Main
SP	Secondary Protected
SSM	Secondary Service Mode Control
SSMI	Secondary Service Mode Indicator
SX	Secondary Extended
UTC	Coordinated Universal Time
VHF	Very High Frequency

3.3 Presentation Conventions

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

- Glossary terms are presented in italics upon their first usage in the text.
- 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.

3.4 Mathematical Symbols

3.4.1 Variable Naming Conventions

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

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 binary vector	Given a binary vector: i = [0, 1, 1, 0, 1, 1, 0, 0] $i_{2:5}$ = [1, 0, 1, 1]

3.4.2 Arithmetic Operators

The arithmetic operators used throughout this document are defined below:

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	$\log_{10}(100) = 2$
| x |	Indicates the absolute value of x	| -5 | = 5 | 3 – 4 | = 1

3.5 FM System Parameters

The FM system parameters used throughout this document are defined below:

Parameter Name	Symbol	Units	Exact Value	Computed Value (To 4 significant figures)
OFDM Subcarrier Spacing	Δf	Hz	1488375/4096	363.4
Cyclic Prefix Width	α	none	7/128	5.469×10^{-2}
OFDM Symbol Duration	T_s	s	$(1 + \alpha) / \Delta f = (135/128)\cdot(4096/1488375)$	2.902×10^{-3}
OFDM Symbol Rate	R_s	Hz	$= 1/T_s$	344.5
L1 Frame Duration	T_f	s	$65536/44100 = 512\cdot T_s$	1.486
L1 Frame Rate	R_f	Hz	$= 1/T_f$	6.729×10^{-1}
L1 Block Duration	T_b	s	$= 32\cdot T_s$	9.288×10^{-2}
L1 Block Rate	R_b	Hz	$= 1/T_b$	10.77
L1 Block Pair Duration	T_p	s	$= 64\cdot T_s$	1.858×10^{-1}
L1 Block Pair Rate	R_p	Hz	$= 1/T_p$	5.383
Digital Diversity Delay Frames	N_{dd}	none	3 = number of L1 frames of diversity delay	3
Digital Diversity Delay Time	T_{dd}	s	$= N_{dd}\cdot T_f$	4.458
Analog Diversity Delay Time	T_{ad}	s	$= 3.0\cdot T_f$	4.458

4 Overview

4.1 Introduction

Layer 1 of the FM system converts information from *Layer 2 (L2)* and *system control* from the *Configuration Administrator* into the FM HD Radio waveform for transmission in the VHF band. Information and control are transported in discrete *transfer frames* via multiple *logical channels*. These transfer frames are also referred to as *Layer 2 Protocol Data Units (PDUs)*.

The L2 PDUs vary in size and format depending on the *service mode*. The service mode, a major component of system control, determines the transmission characteristics of each logical channel. After assessing the requirements of candidate applications, higher protocol layers select service modes that most suitably configure the logical channels. The plurality of logical channels reflects the inherent flexibility of the system, which supports simultaneous delivery of various combinations of digital audio and data.

Layer 1 also receives system control from the Configuration Administrator for use by the Layer 1 System Control Processor.

This section presents the following:

- An overview of the waveforms and spectra
- An overview of the system control, including the available service modes
- An overview of the logical channels
- A high-level discussion of each of the functional components included in the Layer 1 FM air interface

Note: *Throughout this document, various FM system parameters are globally represented as mathematical symbols. Refer to Subsection 3.5 for their values.*

4.2 Waveforms and Spectra

The design provides a flexible means of transitioning to a digital broadcast system by providing three new waveform types: *Hybrid*, *Extended Hybrid*, and *All Digital*. The Hybrid and Extended Hybrid types retain the analog FM signal, while the All Digital type does not. All three waveform types conform to the current *spectral emissions mask*.

The digital signal is modulated using *Orthogonal Frequency Division Multiplexing* (OFDM). OFDM is a parallel modulation scheme in which the data stream modulates a large number of orthogonal subcarriers, which are transmitted simultaneously. OFDM is inherently flexible, readily allowing the mapping of logical channels to different groups of subcarriers.

Refer to Section 5 for a detailed description of the spectra of the three waveform types.

4.2.1 Hybrid Waveform

The digital signal is transmitted in *Primary Main* (PM) *sidebands* on both sides of the analog FM signal in the Hybrid waveform. The power level of each sideband is appreciably below the total power in the analog FM signal. The *analog signal* may be monophonic or stereo and may include Subsidiary Communications Authorization (SCA) channels.

4.2.2 Extended Hybrid Waveform

In the Extended Hybrid waveform, the bandwidth of the Hybrid sidebands can be extended toward the analog FM signal to increase digital capacity. This additional spectrum, allocated to the inner edge of each Primary Main sideband, is termed the *Primary Extended* (PX) *sideband*.

4.2.3 All Digital Waveform

The greatest system enhancements are realized with the All Digital waveform, in which the analog signal is removed, and the bandwidth of the primary digital sidebands is fully extended as in the Extended Hybrid waveform. In addition, this waveform allows lower-power digital *secondary sidebands* to be transmitted in the spectrum vacated by the analog FM signal.

4.3 System Control Channel

The *System Control Channel* (SCCH) transports control and status information. Primary, secondary, and advanced service mode control and amplitude scale factor select are sent from the Configuration Administrator to Layer 1, while synchronization information is sent from Layer 1 to Layer 2. In addition, several bits of the *system control data sequence* designated "reserved" are controlled from layers above L1 via the primary reserved control data interface and the secondary reserved control data interface.

The service mode dictates all permissible configurations of the logical channels. There are two types of service mode: standard and advanced.

- Standard service modes configure primary logical channels and include MP1, MP2, MP3, MP5, MP6, and MP11.
- Advanced service modes include MP1X, DSB1, MP1XOV, DSB1OV, MP6OV, and MS5. All advanced service modes configure primary logical channels except for MS5, which configures secondary logical channels. Some logical channels in advanced service modes use special modulation, coding, and diversity techniques to improve their robustness and/or capacity. Note that MP5OV is not defined because MP6OV provides similar functionality with double the throughput in its most robust (P1) logical channel.

Refer to Section 6 for a detailed description of the SCCH and refer to Section 11 for a detailed description of *System Control Processing*.

4.4 Logical Channels

A logical channel is a signal path that conducts L2 PDUs in transfer frames into Layer 1 with a specific grade of service, determined by service mode. Layer 1 of the FM air interface provides nine logical channels to higher layer protocols. Not all logical channels are used in every service mode. Refer to Subsection 4.4.1 through Subsection 4.4.3 for details.

4.4.1 Primary Logical Channels

There are seven primary logical channels that can be used with the Hybrid, Extended Hybrid, and All Digital waveforms. They are denoted as P1, P2, P3, P4, POV, PIDS, and PIDSOV. The PIDS and PIDSOV channels may transmit the Station Information Service (SIS) information. Table 4-1 shows the approximate information rate supported by each primary logical channel in the standard primary service modes. Table 4-2 shows the approximate information rate supported by each primary logical channel in the advanced primary service modes. Calculations of the exact rates are explained in Section 7.

Table 4-1: Approximate Information Rate of Primary Logical Channels in Standard Primary Service Modes

Service Mode	Approximate Logical Channel Information Rate (kbit/s)					Approximate Total Information Rate (kbit/s)	Waveform
	P1	P2	P3	P4	PIDS		
MP1	98	N/A	N/A	N/A	1	99	Hybrid
MP2	98	N/A	12	N/A	1	111	Extended Hybrid
MP3	98	N/A	25	N/A	1	124	Extended Hybrid
MP11	98	N/A	25	25	1	149	Extended Hybrid
MP5	25	74	25	N/A	1	125	Extended Hybrid, All Digital
MP6	50	49	N/A	N/A	1	100	Extended Hybrid, All Digital

Table 4-2: Approximate Information Rate of Primary Logical Channels in Advanced Primary Service Modes

Service Mode	Approximate Logical Channel Information Rate (kbit/s)						Approximate Total Information Rate (kbit/s)	Waveform
	P1	P2	P4	POV	PIDS	PIDSOV		
MP1X	98	N/A	69	N/A	1	N/A	168	Extended Hybrid
DSB1	229	N/A	N/A	N/A	3	N/A	232	Extended Hybrid, All Digital
MP1XOV	98	N/A	34	114	1	2	249	Extended Hybrid
MP6OV	50	49	N/A	114	1	2	216	Extended Hybrid, All Digital
DSB1OV	229	N/A	N/A	114	3	2	348	Extended Hybrid, All Digital

4.4.2 Secondary Logical Channels

There are two secondary logical channels that are used only with advanced secondary service mode MS5 in the All Digital waveform. They are denoted as S1and SIDS. Table 4-3 shows the approximate information rate supported by each secondary logical channel. Calculations of the exact rates are explained in Section 7.

Table 4-3: Approximate Information Rate of Secondary Logical Channels in Advanced Secondary Service Mode MS5

Service Mode	Approximate Logical Channel Information Rate (kbit/s)		Approximate Total Information Rate (kbit/s)	Waveform
	S1	SIDS		
MS5	114	2	116	All Digital

4.4.3 Logical Channel Functionality

Logical channels P1 through P4, POV, PIDSOV, and S1 are designed to convey audio and data, while the Primary IBOC Data Service (PIDS) and Secondary IBOC Data Service (SIDS) logical channels are designed to carry Station Information Service (SIS) information.

The performance of each logical channel is completely described through three *characterization parameters: transfer, latency,* and *robustness*. *Modulation, channel encoding, spectral mapping, interleaver depth*, and digital *diversity delay* are the components of these characterization parameters. The service mode uniquely configures these components within Layer 1 for each active logical channel, thereby determining the appropriate characterization parameters.

In addition, the service mode specifies the framing and synchronization of the transfer frames through each active logical channel. Refer to Section 7 for a detailed description of the logical channels and their configuration.

4.5 Functional Components

This subsection includes a high-level description of each Layer 1 functional block and the associated signal flow. Figure 4-1 is a functional block diagram of Layer 1 processing.

Some processing stages shown in Figure 4-1 are denoted by a logical channel subscript. For example, logical channel designations are subscripted with an “S” after *scrambling* and with a “G” after channel encoding. In addition, the primed notation (as in $\underline{P1}'_G$) indicates that the logical channel is processed differently than the “unprimed” channel (for example see Figure 9-11 and Figure 9-12) and is destined for transmission in a different portion of the spectrum within the allocated bandwidth. Finally, logical channels P1, P4, PIDS, PIDSOV, POV, S1, and SIDS are denoted with an asterisk following channel encoding, since they are comprised of distinct *main* and *backup* components at that point in the processing in advanced service modes.

The single underline notation for a logical channel name refers to the fact that data is passed between the various functions as *vectors*. Each logical channel has a dedicated scrambler and channel encoder. The configuration administrator is a system function that configures each of the layers using SCCH information or parameters which do not change often. However, dynamic SCCH parameters such as the L1 Block Count and ALFN are sent from Layer 1 to Layer 2.

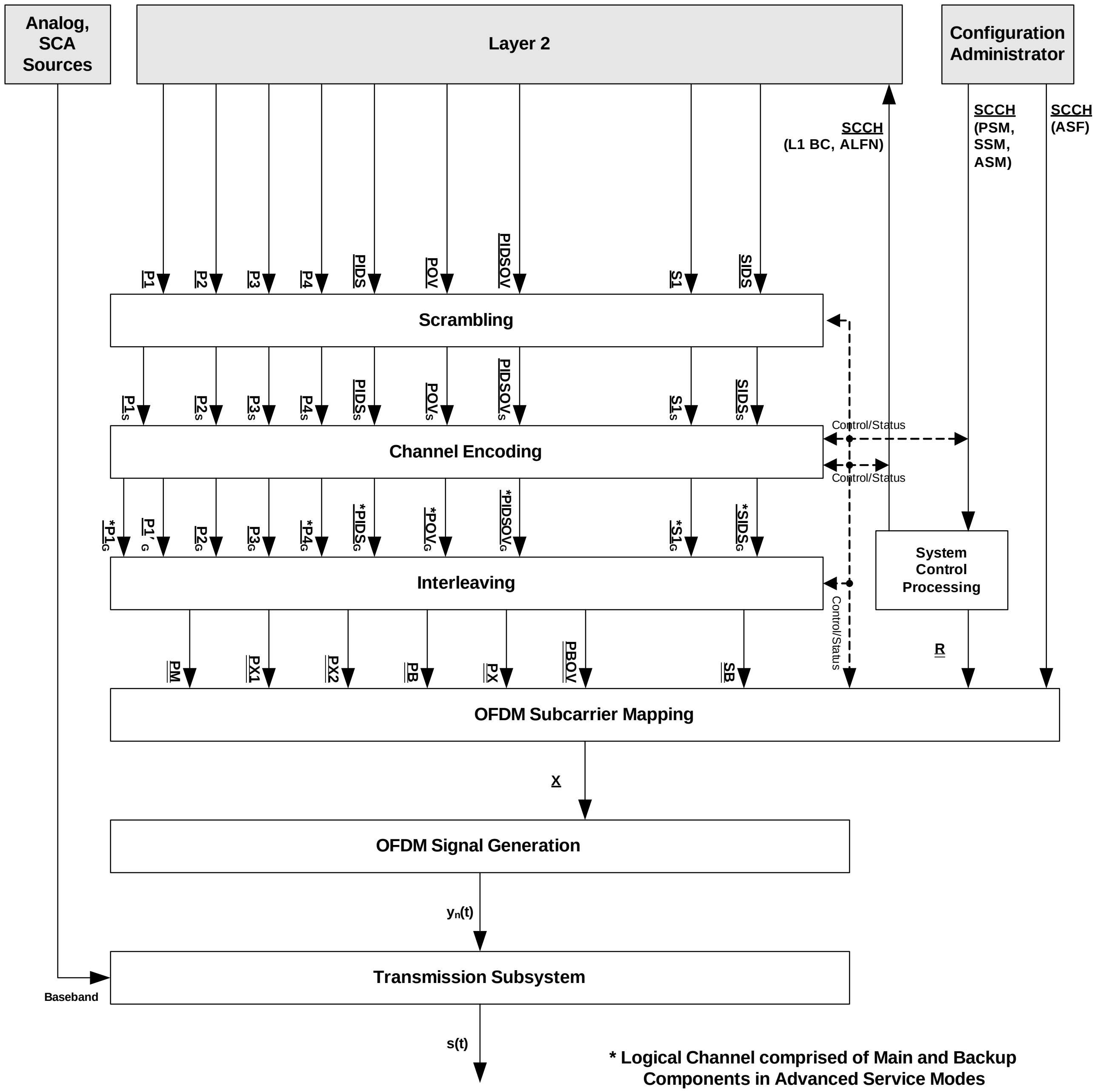


Figure 4-1: FM Air Interface Layer 1 Functional Block Diagram

NOTE | Figure 4-1 has been updated since NRSC-5-D

4.5.1 Scrambling

This function randomizes the digital data in each logical channel to mitigate signal periodicities. At the output of the scrambling function, the logical channel vectors retain their identity, but are distinguished by the “S” subscript (e.g., “$\underline{P1}_S$”). Refer to Section 8 for a detailed description of the scrambling functional component.

4.5.2 Channel Encoding

This function uses *convolutional encoding* to add redundancy to the digital data in each logical channel to improve its reliability in the presence of channel impairments. The size of the logical channel vectors is increased in inverse proportion to the *code rate*. The encoding techniques are configurable by service mode. Digital Diversity delay is also imposed on selected logical channels. At the output of the channel encoder, the logical channel vectors retain their identity, but are distinguished now by the “G” subscript (e.g., “$\underline{P1}_G$”). In advanced service modes, some logical channels are split into main and backup components to provide additional robustness via time diversity; they are distinguished by a B or M superscript (e.g., “$\underline{P1}^B$”). In a few service modes, P1 and S1 are split to provide a delayed and undelayed version at the output. Refer to Section 9 for a detailed description of the channel encoding functional component.

4.5.3 Interleaving

Interleaving in time and frequency is employed to mitigate the effects of burst errors. The interleaving techniques are tailored to the VHF *fading* environment and are configurable by service mode. In this process, the logical channels lose their identity. The interleaver output is structured in a matrix format; each matrix consists of one or more logical channels and is associated with a particular portion of the transmitted spectrum. The *interleaver matrix* designations reflect the spectral mapping. For example, “$\underline{\underline{PM}}$” maps to the Primary Main portion of the spectrum, “$\underline{\underline{PX1}}$” maps to the Primary Extended (PX) portion of the spectrum, and “$\underline{\underline{SB}}$”(secondary broadband) maps to the entire secondary portion of the spectrum. In advanced service modes, some logical channels from the channel encoder are presented to separate main and backup interleavers, with diversity delay applied to the backup component to improve robustness. Refer to Section 10 for a detailed description of the interleaving functional component.

4.5.4 System Control Processing

This function generates a matrix of system control data sequences that include control and status (such as service mode), for broadcast on the *reference subcarriers*. This data matrix is designated “$\underline{\underline{R}}$” for “Reference.” Refer to Section 11 for a detailed description of the system control processing functional component.

4.5.5 OFDM Subcarrier Mapping

This function assigns the interleaver matrices and the system control matrix to the *OFDM subcarriers*. One row of each active interleaver matrix is processed every *OFDM symbol* T_s to produce one output vector $\underline{X}$ which is a frequency-domain representation of the signal. The mapping is specifically tailored to the non-uniform interference environment and is a function of the service mode. In standard service modes, QPSK modulation is applied to each OFDM data subcarrier. However, in advanced service modes, the data subcarriers may be modulated using QPSK (two bits per subcarrier), *16 QAM* (four bits per subcarrier), or *64 QAM* (six bits per subcarrier) to enhance the capacity of the waveform. Refer to Section 12 for a detailed description of the *OFDM Subcarrier Mapping* functional component.

4.5.6 OFDM Signal Generation

This function generates the digital portion of the time-domain FM HD Radio waveform. The input vectors are transformed into a shaped time-domain baseband pulse, $y_n(t)$, defining one OFDM symbol. Refer to Section 13 for a detailed description of the *OFDM Signal Generation* functional component.

4.5.7 Transmission Subsystem

This function formats the baseband waveform for transmission through the VHF channel. Major sub-functions include symbol concatenation and frequency up-conversion. In addition, when transmitting the Hybrid waveform, this function modulates the analog source and combines it with the digital signal to form a composite Hybrid signal, s(t), ready for transmission. Refer to Section 14 for a detailed description of the *transmission subsystem* functional component.

5 Waveforms and Spectra

5.1 Introduction

This section describes the output spectrum for each of the three digital waveform types: Hybrid, Extended Hybrid, and All Digital. Each spectrum is divided into several sidebands which represent various subcarrier groupings. All spectra are represented at baseband.

5.2 Frequency Partitions and Spectral Conventions

The OFDM subcarriers are assembled into *frequency partitions*. Each frequency partition consists of eighteen data subcarriers and one reference subcarrier as shown in Figure 5-1 (Ordering A) and Figure 5-2 (Ordering B). The position of the reference subcarrier (Ordering A or B) varies with the location of the frequency partition within the spectrum.

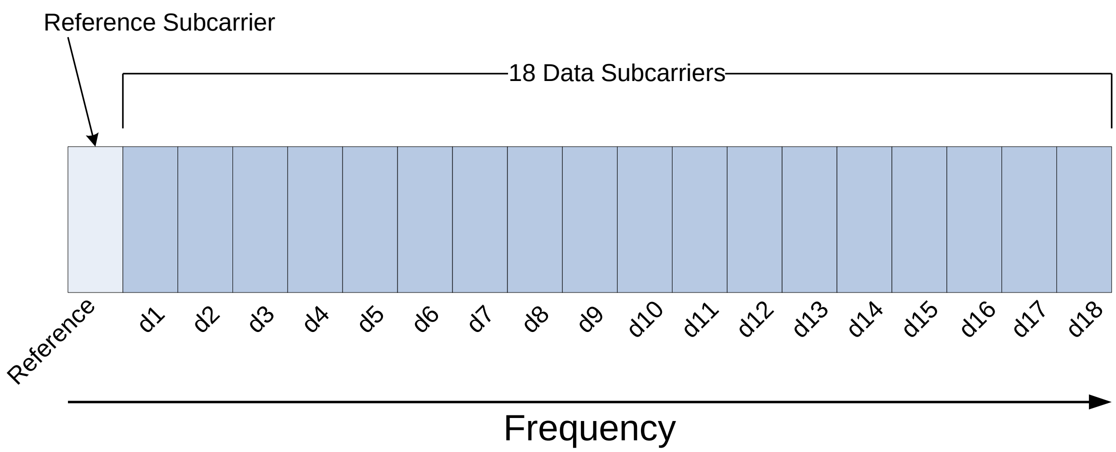


Figure 5-1: Frequency Partition – Ordering A

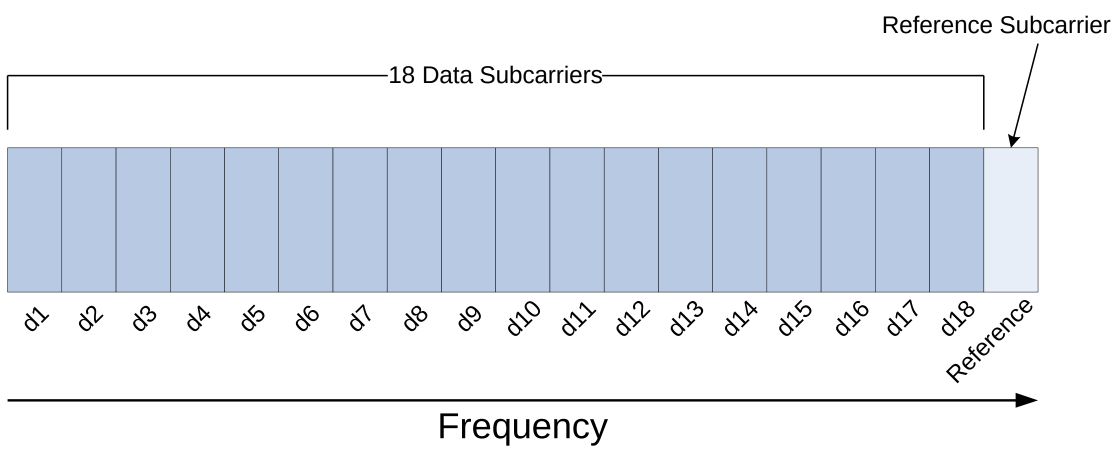


Figure 5-2: Frequency Partition – Ordering B

For each frequency partition, data subcarriers d1 through d18 convey the payload (data or encoded audio) from Layer 2 while the reference subcarriers convey L1 system control. Subcarriers are numbered from minus 546 at the lower end to zero at the center frequency to plus 546 at the upper end of the channel frequency allocation.

Besides the reference subcarriers resident within each frequency partition, depending on the service mode, up to four additional reference subcarriers are inserted into the spectrum at the following subcarrier numbers: -546, -1, +1, and +546. The overall effect is a regular distribution of reference subcarriers throughout the spectrum. For notational convenience, each reference subcarrier is assigned a unique identification number between 0 and 60. All lower sideband reference subcarriers are shown in Figure 5-3. All upper sideband reference subcarriers are shown in Figure 5-4. The figures indicate the relationship between reference subcarrier numbers and OFDM subcarrier numbers. Note that there is no reference subcarrier 30, as subcarrier 0 is unused.

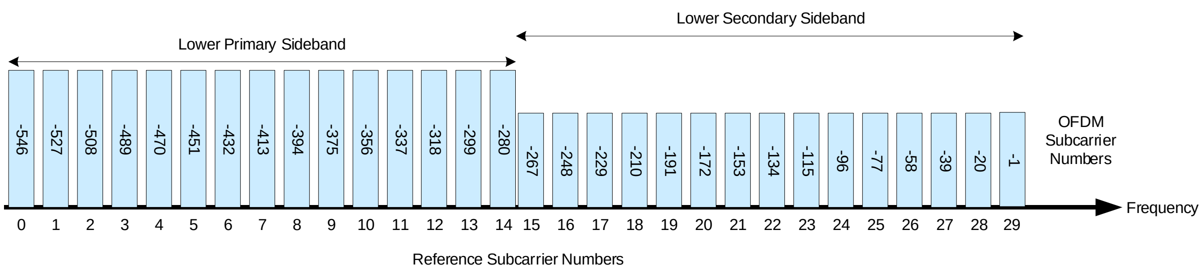


Figure 5-3: Lower Sideband Reference Subcarrier Spectral Mapping

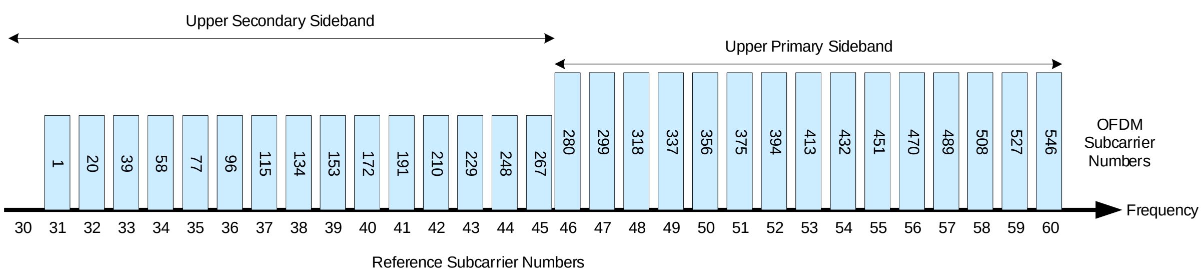


Figure 5-4: Upper Sideband Reference Subcarrier Spectral Mapping

Each spectrum described in the remaining subsections shows the subcarrier number and center frequency of certain key OFDM subcarriers. The center frequency of a subcarrier is calculated by multiplying the subcarrier number by the OFDM subcarrier spacing Δf. The center of subcarrier 0 is located at 0 Hz. In this context, center frequency is relative to the radio frequency (RF) *allocated channel*.

For example, the upper Primary Main sideband is bounded by subcarriers 356 and 546 whose center frequencies are located at 129,361 Hz and 198,402 Hz, respectively. The frequency span of a Primary Main sideband is 69,041 Hz (198,402 Hz – 129,361 Hz).

5.3 Hybrid Spectrum

The digital signal is transmitted in PM sidebands on both sides of the analog FM signal as shown in Figure 5-5. Each PM sideband consists of ten frequency partitions which are allocated among subcarriers 356 through 545, or -356 through -545. Subcarriers 546 and -546, also included in the PM sidebands, are additional reference subcarriers. The amplitude of each subcarrier is scaled by an *amplitude scale factor* as indicated in Table 5-1. All of the subcarriers within the lower sideband share a common scale factor, a_{0L}, so that these subcarriers have the same amplitude relative to one another. Similarly, all of the subcarriers within the upper sideband share a common scale factor, a_{0U}, so that these subcarriers have the same amplitude relative to one another. However, a_{0L} and a_{0U} may be different; the upper and lower sidebands may differ in average power level by up to 10 dB (asymmetric sidebands). Normally, the sideband power levels are equal, but under certain scenarios, asymmetric sidebands may be useful for mitigation of adjacent channel interference. Refer to [7] for further details.

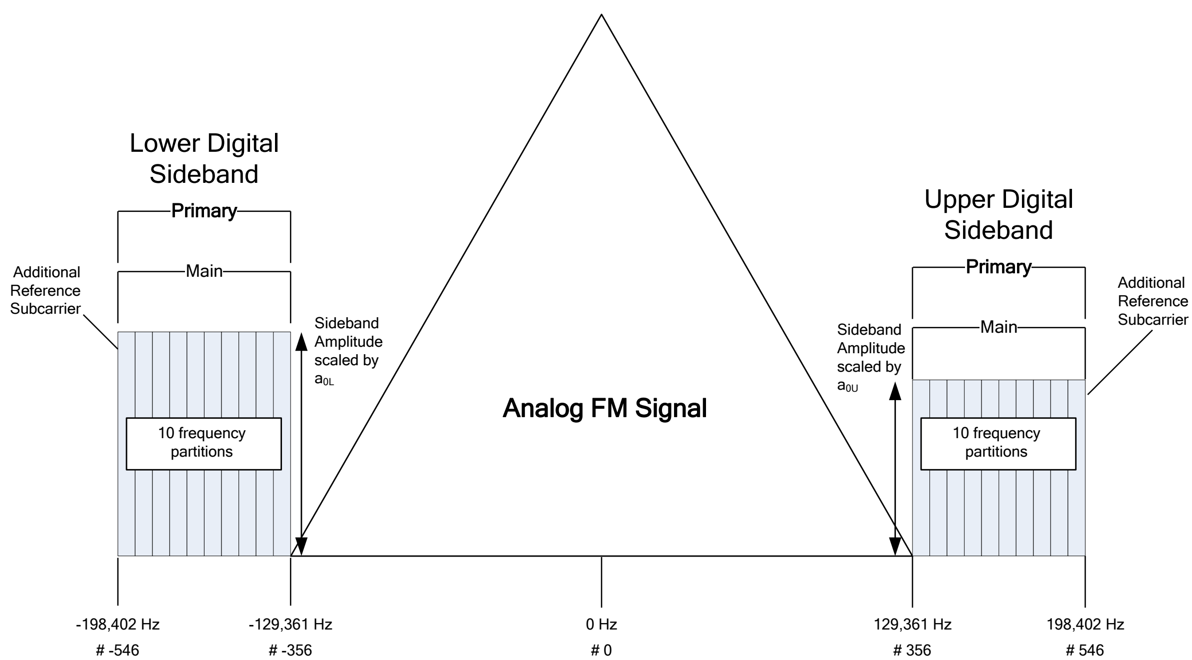


Figure 5-5: Spectrum of the Hybrid Waveform – Service Mode MP1

Table 5-1 summarizes the upper and lower Primary Main sidebands for the Hybrid waveform.

Table 5-1: Hybrid Waveform Spectral Summary – Service Mode MP1

Sideband	Number of Frequency Partitions	Frequency Partition Ordering	Subcarrier Range	Subcarrier Frequencies (Hz from channel center)	Frequency Span (Hz)	Amplitude Scale Factor	Comments
Upper Primary Main	10	A	356 to 546	129,361 to 198,402	69,041	a_{0U}	Includes additional reference subcarrier located at subcarrier 546
Lower Primary Main	10	B	-356 to -546	-129,361 to -198,402	69,041	a_{0L}	Includes additional reference subcarrier located at subcarrier -546

Note: *Refer to Reference [7] for details regarding the amplitude scale factors shown above.*

5.4 Extended Hybrid Spectrum

The Extended Hybrid waveform is created by adding Primary Extended sidebands to the Primary Main sidebands present in the Hybrid waveform as shown in Figure 5-6. Depending on the service mode, one, two, or four frequency partitions can be added to the inner edge of each Primary Main sideband.

Each Primary Main sideband consists of ten frequency partitions and an additional reference subcarrier spanning subcarriers 356 through 546, or -356 through -546. The upper Primary Extended sidebands include subcarriers 337 through 355 (one frequency partition), 318 through 355 (two frequency partitions), or 280 through 355 (four frequency partitions). The lower Primary Extended sidebands include subcarriers -337 through -355 (one frequency partition), -318 through -355 (two frequency partitions), or -280 through -355 (four frequency partitions).

The amplitude of each subcarrier is scaled by an amplitude scale factor as indicated in Table 5-2. All of the subcarriers within the lower sideband share a common scale factor, a_{0L}, so that these subcarriers have the same amplitude relative to one another. Similarly, all of the subcarriers within the upper sideband share a common scale factor, a_{0U}, so that these subcarriers have the same amplitude relative to one another. However, a_{0L} and a_{0U} may be different; the upper and lower sidebands may differ in average power level by up to 10 dB (asymmetric sidebands). Normally, the sideband power levels are equal, but under certain scenarios, asymmetric sidebands may be useful for mitigation of adjacent channel interference. Refer to [7] for further details.

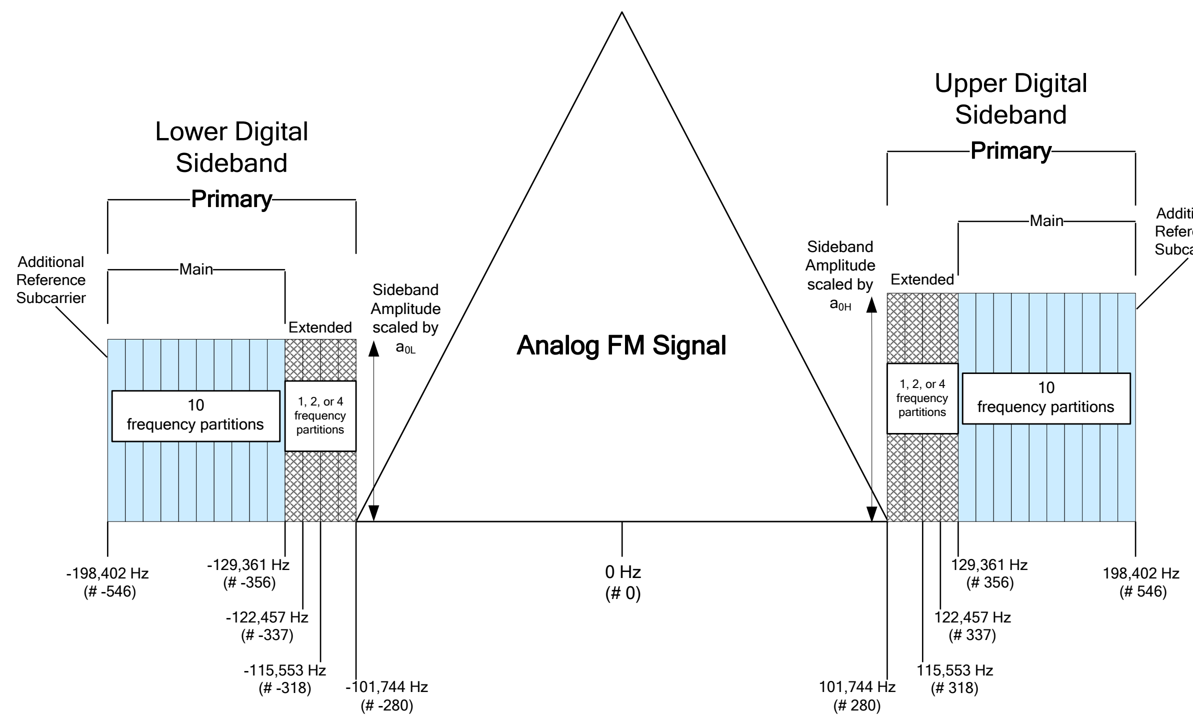


Figure 5-6: Spectrum of the Extended Hybrid Waveform – Service Modes MP2, MP3, MP11, MP5, MP6, MP1X, DSB1, MP1XOV, MP6OV, and DSB1OV

Table 5-2 summarizes the Upper and Lower Primary sidebands for the Extended Hybrid waveform.

Table 5-2: Extended Hybrid Waveform Spectral Summary – Service Modes MP2, MP3, MP11, MP5, MP6, MP1X, DSB1, MP1XOV, MP6OV, and DSB1OV

Sideband	Number Of Frequency Partitions	Frequency Partition Ordering	Subcarrier Range	Subcarrier Frequencies (Hz from channel center)	Freq. Span (Hz)	Ampl. Scale Factor	Comments
Upper Primary Main	10	A	356 to 546	129,361 to 198,402	69,041	a_{0U}	Includes additional reference subcarrier located at subcarrier 546
Lower Primary Main	10	B	-356 to -546	-129,361 to -198,402	69,041	a_{0L}	Includes additional reference subcarrier located at subcarrier -546
Upper Primary Extended (1 frequency partition)	1	A	337 to 355	122,457 to 128,997	6,540	a_{0U}	none
Lower Primary Extended (1 frequency partition)	1	B	-337 to -355	-122,457 to -128,997	6,540	a_{0L}	none
Upper Primary Extended (2 frequency partitions)	2	A	318 to 355	115,553 to 128,997	13,444	a_{0U}	none
Lower Primary Extended (2 frequency partitions)	2	B	-318 to -355	-115,553 to -128,997	13,444	a_{0L}	none
Upper Primary Extended (4 frequency partitions)	4	A	280 to 355	101,744 to 128,997	27,253	a_{0U}	none

Sideband	Number Of Frequency Partitions	Frequency Partition Ordering	Subcarrier Range	Subcarrier Frequencies (Hz from channel center)	Freq. Span (Hz)	Ampl. Scale Factor	Comments
Lower Primary Extended (4 frequency partitions)	4	B	-280 to -355	-101,744 to -128,997	27,253	a_{0L}	none

Note: *Refer to Reference [7] for details regarding the amplitude scale factors shown above.*

5.5 All Digital Spectrum

The All Digital waveform is constructed by disabling the analog signal, fully expanding the bandwidth of the primary digital sidebands, and adding lower-power secondary sidebands in the spectrum vacated by the analog signal. The spectrum of the All Digital waveform is shown in Figure 5-7.

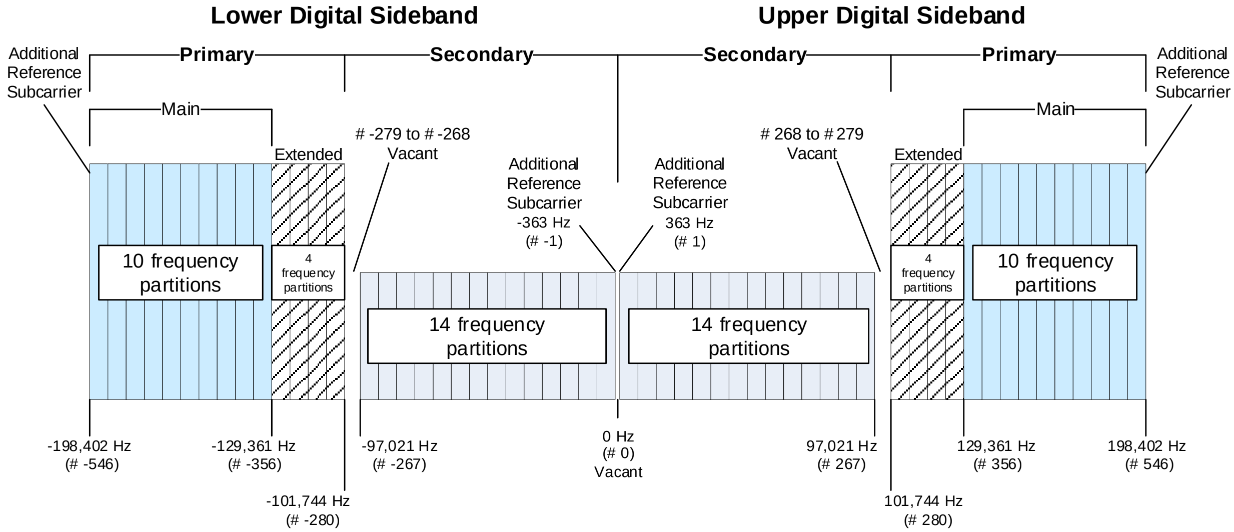


Figure 5-7: Spectrum of the All Digital Waveform – Service Modes MP5, MP6, DSB1, MP6OV, DSB1OV, and MS5

In addition to the ten main frequency partitions, all four extended frequency partitions are present in each primary sideband of the All Digital waveform. Each secondary sideband also has fourteen frequency partitions.

The lower secondary sideband includes subcarriers -267 through -1, and the upper secondary sideband includes subcarriers 1 through 267. Lower secondary subcarriers -279 through -268 and upper secondary subcarriers 268 through 279, as well as subcarrier 0, are not populated. Additional secondary reference subcarriers are placed near the center of the channel, at subcarriers -1 and 1. The total frequency span of the entire All Digital spectrum is 396,803 Hz.

The amplitude of each subcarrier is scaled by an amplitude scale factor, as indicated in Table 5-3. All subcarriers within the lower primary, upper primary, lower secondary, or upper secondary sideband share a common scale factor, so that these subcarriers have the same amplitude relative to one another. However, as with hybrid and extended hybrid waveforms, the upper and lower scale factors may be different (asymmetric sidebands). Normally, the sideband power levels are equal, but under certain scenarios, asymmetric sidebands may be useful for mitigation of adjacent channel interference. Refer to [7] for further details.

The secondary sideband amplitude scale factors, a_{2U} through a_{5U} and a_{2L} through a_{5L}, are user selectable. Any one of the four may be selected for application to an upper or lower secondary sideband. Table 5-3 summarizes the upper and lower, primary and secondary sidebands for the All Digital waveform.

Table 5-3: All Digital Waveform Spectral Summary – Service Modes MP5, MP6, DSB1, MP6OV, DSB1OV, and MS5

Sideband	Number Of Frequency Partitions	Freq. Partition Ordering	Subcarrier Range	Subcarrier Frequencies (Hz from channel center)	Freq. Span (Hz)	Ampl. Scale Factor	Comments

Sideband	Number Of Frequency Partitions	Freq. Partition Ordering	Subcarrier Range	Subcarrier Frequencies (Hz from channel center)	Freq. Span (Hz)	Ampl. Scale Factor	Comments
Upper Primary Main	10	A	356 to 546	129,361 to 198,402	69,041	a_{1U}	Includes additional reference subcarrier located at subcarrier 546
Lower Primary Main	10	B	-356 to -546	-129,361 to -198,402	69,041	a_{1L}	Includes additional reference subcarrier located at subcarrier -546
Upper Primary Extended	4	A	280 to 355	101,744 to 128,997	27,253	a_{1U}	none
Lower Primary Extended	4	B	-280 to -355	-101,744 to -128,997	27,253	a_{1L}	none
Upper Secondary	14	B	1 to 267	363 to 97,021	96,658	a_{2U}, a_{3U}, a_{4U}, a_{5U}	Includes additional reference subcarrier located at subcarrier 1
Lower Secondary	14	A	-1 to -267	-363 to -97,021	96,658	a_{2L}, a_{3L}, a_{4L}, a_{5L}	Includes additional reference subcarrier located at subcarrier -1

Note: *Refer to Reference [7] for details regarding the amplitude scale factors shown above and Subsection 6.6 for information on how a_{2U} – a_{5U} and a_{2L} – a_{5L} are selected.*

6 System Control Channel

6.1 Introduction

The SCCH passes discrete transfer frames of control and status information among Layer 2, the Configuration Administrator, and Layer 1. The control information passed from the Configuration Administrator to Layer 1 consists of Primary Service Mode Control (PSM), Secondary Service Mode Control (SSM), Advanced Service Mode Control (ASM), Primary Amplitude Scale Factors (a_{0L}, a_{0U}, a_{1L}, and a_{1U}), and Secondary Amplitude Scale Factor Select (ASF). ASF selects from several fixed scale factors to set the levels of the upper and lower secondary sidebands. The Primary Amplitude Scale Factors are actual values sent from the Configuration Administrator to establish the absolute power levels of the primary sidebands. The update rate and resolution of the Primary Amplitude Scale Factors, and the actual point or points within the signal path where they are applied depends on the specific implementation and is outside the scope of this document. The status information passed from Layer 1 to Layer 2 consists of *Absolute L1 Frame Number* (ALFN) and *L1 Block Count* (BC). In addition, several bits of the system control data sequence designated "reserved" are controlled by the Configuration Administrator. Refer to Figure 6-1. This status information, the L1 Block Count, and indicators of the state of the control information (with the exception of ALFN) are broadcast on the reference subcarriers.

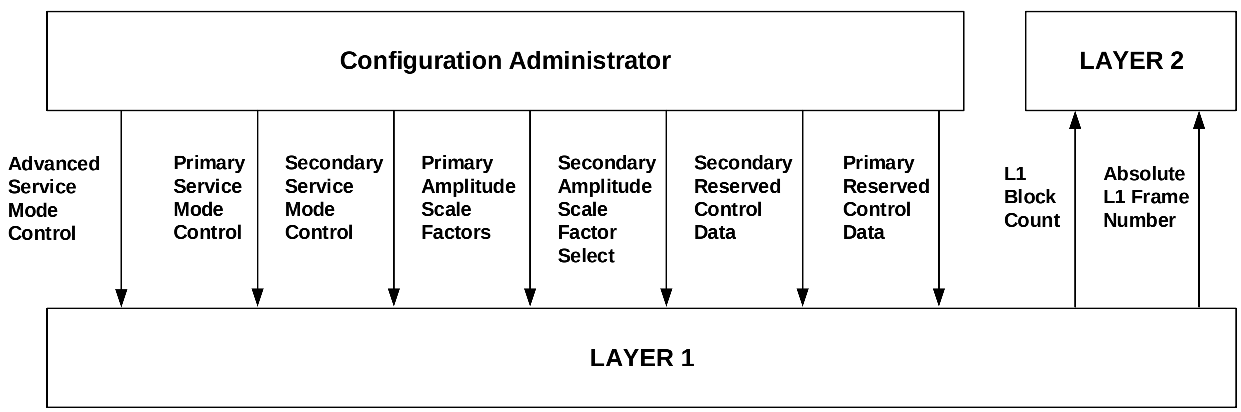


Figure 6-1: System Control Channel

The direction and rate of transfer between Layer 2, the Configuration Administrator, and Layer 1 is given in Table 6-1.

Table 6-1: Transfer through the System Control Channel (SCCH)

Data	Direction	Transfer Frame Rate	Size (bits)
Advanced Service Mode Control (ASM)	Configuration Administrator → Layer 1	R_f	2
Primary Service Mode Control (PSM)	Configuration Administrator → Layer 1	R_f	6
Secondary Service Mode Control (SSM)	Configuration Administrator → Layer 1	R_f	3

Data	Direction	Transfer Frame Rate	Size (bits)
Primary Amplitude Scale Factors (a_{0L}, a_{0U}, a_{1L}, and a_{1U} only)	Configuration Administrator → Layer 1	N/A	N/A
Secondary Amplitude Scale Factor Select (ASF)	Configuration Administrator → Layer 1	R_f	4
Primary Reserved Control Data	Configuration Administrator → Layer 1	R_f	2
Secondary Reserved Control Data	Configuration Administrator → Layer 1	R_f	8
L1 Block Count (BC)	Layer 1 → Layer 2	R_b	4
Absolute L1 Frame Number (ALFN)	Layer 1 → Layer 2	R_f	32

6.2 Service Mode Control

The service mode dictates the configuration and performance of the logical channels. Primary service modes configure primary logical channels and secondary service modes configure secondary logical channels.

All waveforms require the definition of both primary and secondary service modes. If secondary sidebands are not present, the secondary service mode is set to "None" as shown in Table 6-3. The service modes support the delivery of various combinations of digital audio and data.

- The active primary service modes defined by this document are MP1, MP2, MP3, MP11, MP5, MP6, MP1X, DSB1, MP6OV, MP1XOV, and DSB1OV.
- The active secondary service mode defined by this document is MS5.

Table 6-2 and Table 6-3 define the bit mapping for PSM and SSM, respectively.

Table 6-2: PSM Bit Mapping

Primary Service Mode	**Bit Assignment (bits 5:0)**					
	5	**4**	**3**	**2**	**1**	**0**
DSB1, DSB1OV	0	0	0	0	0	0
MP1	0	0	0	0	0	1
MP2	0	0	0	0	1	0
MP3	0	0	0	0	1	1
Reserved	0	0	0	1	0	0
MP5	0	0	0	1	0	1
MP6	0	0	0	1	1	0
Reserved	0	0	0	1	1	1
…	…	…	…	…	…	…
Reserved	0	0	1	0	1	0
MP11	0	0	1	0	1	1
Reserved	0	0	1	1	0	0
Reserved	0	0	1	1	0	1
MP6OV	0	0	1	1	1	0
Reserved	0	0	1	1	1	1
…	…	…	…	…	…	…
Reserved	0	1	0	0	1	1
MP1X	0	1	0	1	0	0
Reserved	0	1	0	1	0	1
…	…	…	…	…	…	…
Reserved	0	1	1	0	0	0
MP1XOV	0	1	1	0	0	1
Reserved	0	1	1	0	1	0
…	…	…	…	…	…	…
Reserved	1	1	1	1	1	1

Table 6-3: SSM Bit Mapping

Secondary Service Mode	**Bit Assignment (bits 2:0)**		
	2	**1**	**0**
None	0	0	0
Reserved	0	0	1
Reserved	0	1	0
Reserved	0	1	1
Reserved	1	0	0
MS5	1	0	1
Reserved	1	1	0
Reserved	1	1	1

In Table 6-2, two advanced primary service modes (DSB1 and DSB1OV) are defined by primary service mode 0 because they are not backward compatible with any standard service mode. Receivers configured for only standard service modes will not acquire signals configured by these service modes, because primary service mode 0 is not defined. However, receivers configured for advanced service modes will recognize that primary service mode 0 signifies one of DSB1 or DSB1OV.

In the near term, non-backward-compatible advanced service modes could be used to broadcast specialized applications (e.g., new data services) that do not require backward compatibility with existing car receivers. Eventually, when enough HD Radio receivers in the market support these service modes, radio stations could use them to broadcast any desired content.

The Advanced Service Mode Control (ASM) specifies which of the non-backward-compatible advanced service modes is configuring the transmitted waveform when the primary service mode is 0. It is comprised of bits 8 and 19 from the 32-bit primary system control data sequence. These two bits have fixed values unless the primary service mode is 0. The Advanced Service Mode Control bit mapping is defined in Table 6-4. Refer to Section 11 for a detailed description of *System Control Processing*.

Table 6-4: Advanced Service Mode Control Bit Mapping

Advanced Primary Service Mode	Primary System Control Data Sequence Bit 19	Primary System Control Data Sequence Bit 8
DSB1	0	0
Reserved	0	1
DSB1OV	1	0
Reserved	1	1

6.2.1 Primary Service Mode Backward Compatibility

Reserved primary service mode bit assignments are for future expansion. To ensure backward compatibility, all primary service modes defined as “Reserved” in Table 6-2 must maintain backward compatibility with one of the following service modes: MP1, MP2, MP3, MP11, MP5, or MP6.

As a minimum, backward compatibility includes the PIDS logical channel, the system control data sequence (matrix $\underline{\underline{R}}$) conveyed over the reference subcarriers, and at least one logical channel which can support medium-quality digital audio. Refer to Table 6-5 for a definition of the default service modes that first generation receivers will assume and with which all transmission equipment must maintain backward compatibility for all reserved primary service mode assignments. Any service mode that is backward compatible with Hybrid service modes MP1-MP3 (e.g., MP9, MP10, MP19, and MP28) is also a Hybrid service mode and the secondary service mode must be set to “None”.

A primary service mode may maintain backward compatibility with primary service modes MP5 and MP6 in one of two configurations. Both the P1 and P1′ or only the P1′ logical channels may be supported. For each primary service mode, Table 6-5 defines which logical channels must maintain backward compatibility.

MP11 is a special case. First-generation receivers will fall back to service mode MP3 and will decode the P3 logical channel and ignore the P4 logical channel. However, MP11 is fully defined in this document and is no longer reserved.

Several advanced service modes are also fully defined in this document and are no longer reserved. Advanced primary service modes MP1X (mapped to MP20) and MP1XOV (mapped to MP25) are backward compatible with MP1; first-generation receivers will decode the P1 logical channel, but will ignore the P4, POV, and PIDSOV logical channels. Advanced primary service mode MP6OV (mapped to MP14) is backward compatible with MP6, so first-generation receivers will decode P1 and P2, but will ignore the POV and PIDSOV logical channels.

As described above, advanced primary service modes DSB1 and DSB1OV are not backward compatible and are mapped to primary service mode 0 (MP0), which cannot be acquired by receivers equipped for only standard service modes. Properly configured receivers, however, will be able to acquire and decode these signals.

Table 6-5: Reserved Primary Service Modes – Defaults

Actual Primary Service Mode	**Bit Assignment (bits 5:0)**						**Default Primary Service Mode**	**Backward Compatible Logical Channels/Elements**
	5	**4**	**3**	**2**	**1**	**0**		
MP4	0	0	0	1	0	0	MP1	P1, PIDS, R, Analog
MP7	0	0	0	1	1	1	MP5	P1′, PIDS, R
MP8	0	0	1	0	0	0	MP6	P1′, PIDS, R
MP9	0	0	1	0	0	1	MP1	P1, PIDS, R, Analog
MP10	0	0	1	0	1	0	MP2	P1, PIDS, R, Analog
MP11	0	0	1	0	1	1	MP3	P1, P3, PIDS, R, Analog
MP12	0	0	1	1	0	0	MP1	P1, PIDS, R, Analog
MP13	0	0	1	1	0	1	MP5	P1, P1′, PIDS, R
MP14 (MP6OV)	0	0	1	1	1	0	MP6	P1, P1′, PIDS, R
MP15	0	0	1	1	1	1	MP5	P1′, PIDS, R
MP16	0	1	0	0	0	0	MP6	P1′, PIDS, R
MP17	0	1	0	0	0	1	MP1	P1, PIDS, R, Analog
MP18	0	1	0	0	1	0	MP2	P1, PIDS, R, Analog
MP19	0	1	0	0	1	1	MP3	P1, P3, PIDS, R, Analog
MP20 (MP1X)	0	1	0	1	0	0	MP1	P1, PIDS, R, Analog
MP21	0	1	0	1	0	1	MP5	P1, P1′, PIDS, R
MP22	0	1	0	1	1	0	MP6	P1, P1′, PIDS, R
MP23	0	1	0	1	1	1	MP5	P1′, PIDS, R
MP24	0	1	1	0	0	0	MP6	P1′, PIDS, R
MP25 (MP1XOV)	0	1	1	0	0	1	MP1	P1, PIDS, R, Analog
MP26	0	1	1	0	1	0	MP2	P1, PIDS, R, Analog
MP27	0	1	1	0	1	1	MP11	P1, P3, P4, PIDS, R, Analog
MP28	0	1	1	1	0	0	MP1	P1, PIDS, R, Analog
MP29	0	1	1	1	0	1	MP5	P1, P1′, PIDS, R
MP30	0	1	1	1	1	0	MP6	P1, P1′, PIDS, R
MP31	0	1	1	1	1	1	MP5	P1′, PIDS, R
MP32	1	0	0	0	0	0	MP6	P1′, PIDS, R
MP33	1	0	0	0	0	1	MP1	P1, PIDS, R, Analog
MP34	1	0	0	0	1	0	MP2	P1, PIDS, R, Analog
MP35	1	0	0	0	1	1	MP3	P1, P3, PIDS, R, Analog
MP36	1	0	0	1	0	0	MP1	P1, PIDS, R, Analog
MP37	1	0	0	1	0	1	MP5	P1, P1′, PIDS, R
MP38	1	0	0	1	1	0	MP6	P1, P1′, PIDS, R
MP39	1	0	0	1	1	1	MP5	P1′, PIDS, R
MP40	1	0	1	0	0	0	MP6	P1′, PIDS, R
MP41	1	0	1	0	0	1	MP1	P1, PIDS, R, Analog
MP42	1	0	1	0	1	0	MP2	P1, PIDS, R, Analog
MP43	1	0	1	0	1	1	MP11	P1, P3, P4, PIDS, R, Analog

Actual Primary Service Mode	Bit Assignment (bits 5:0)						Default Primary Service Mode	Backward Compatible Logical Channels/Elements
	5	4	3	2	1	0		
MP44	1	0	1	1	0	0	MP1	P1, PIDS, R, Analog
MP45	1	0	1	1	0	1	MP5	P1, P1′, PIDS, R
MP46	1	0	1	1	1	0	MP6	P1, P1′, PIDS, R
MP47	1	0	1	1	1	1	MP5	P1′, PIDS, R
MP48	1	1	0	0	0	0	MP6	P1′, PIDS, R
MP49	1	1	0	0	0	1	MP1	P1, PIDS, R, Analog
MP50	1	1	0	0	1	0	MP2	P1, PIDS, R, Analog
MP51	1	1	0	0	1	1	MP3	P1, P3, PIDS, R, Analog
MP52	1	1	0	1	0	0	MP1	P1, PIDS, R, Analog
MP53	1	1	0	1	0	1	MP5	P1, P1′, PIDS, R
MP54	1	1	0	1	1	0	MP6	P1, P1′, PIDS, R
MP55	1	1	0	1	1	1	MP5	P1′, PIDS, R
MP56	1	1	1	0	0	0	MP6	P1′, PIDS, R
MP57	1	1	1	0	0	1	MP1	P1, PIDS, R, Analog
MP58	1	1	1	0	1	0	MP2	P1, PIDS, R, Analog
MP59	1	1	1	0	1	1	MP11	P1, P3, P4, PIDS, R, Analog
MP60	1	1	1	1	0	0	MP1	P1, PIDS, R, Analog
MP61	1	1	1	1	0	1	MP5	P1, P1′, PIDS, R
MP62	1	1	1	1	1	0	MP6	P1, P1′, PIDS, R
MP63	1	1	1	1	1	1	MP5	P1′, PIDS, R

6.2.2 Service Mode Pairings

When broadcasting secondary sidebands in the All Digital waveform, active primary and secondary service modes are both required. Any Hybrid-only or Extended-Hybrid-only primary service modes are invalid for the All Digital waveform (e.g., MP1, MP2, MP3, MP11, MP1X, and MP1XOV). Only primary service modes MP5, MP6, DSB1, MP6OV, and DSB1OV may be paired with secondary service mode MS5 when broadcasting the All Digital waveform. Any combination of these primary and secondary service modes is allowable.

6.2.3 Service Mode Switching

Primary service mode control (PSM) and secondary service mode control (SSM) are received from the Configuration Administrator via the SCCH at the rate R_f. Service mode changes are invoked only on an *L1 frame* boundary (see Subsection 6.3).

6.3 Absolute L1 Frame Number (ALFN)

The transmitted HD Radio signal may be regarded as a series of unique L1 frames of duration T_f. In order to reference all transmissions to absolute time, each L1 frame is associated with an ALFN. This universal frame numbering scheme assumes that the start of ALFN 0 occurred at 00:00:00 Coordinated Universal Time (UTC) on January 6, 1980. The start of every subsequent L1 frame occurs at an exact integer multiple of T_f after that instant in time. The current ALFN is a binary number determined by subtracting the GPS start time (00:00:00 on January 6, 1980) from the current GPS time (making allowance for the GPS epoch), expressing the difference in seconds, and multiplying the result by the frame rate R_f.

The ALFN (which is passed to Layer 2 via the SCCH at the frame rate R_f) may be used to schedule the delivery of time-critical programming.

6.4 L1 Block Count

Each L1 frame may be considered to consist of sixteen *L1 blocks* of duration T_b. The L1 Block Count (BC) indicates the position of the current L1 block within the L1 frame. An L1 block count of zero signifies the start of an L1 frame, while a BC of 15 designates the final L1 block in an L1 frame. Table 6-6 defines the L1 BC bit mapping.

The BC is passed to Layer 2 via the SCCH at the block rate R_b. It is broadcast on the reference subcarriers and is used by the receiver to aid in synchronization.

An illustration of the relationship of L1 blocks to L1 frames is shown in Figure 6-2.

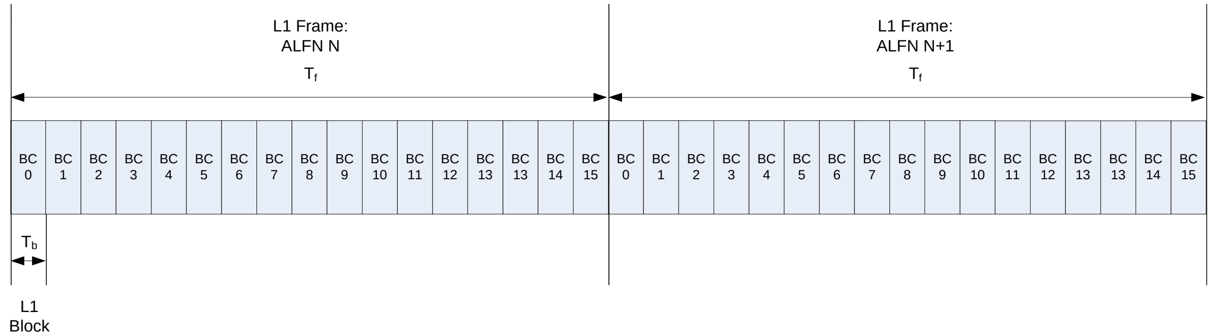


Figure 6-2: L1 Frames and L1 Blocks

Table 6-6: L1 Block Count (BC) Bit Mapping

L1 Block Count	Bit Assignment (bits 3:0)			
	3	2	1	0
0	0	0	0	0
1	0	0	0	1
2	0	0	1	0
3	0	0	1	1
4	0	1	0	0
5	0	1	0	1
6	0	1	1	0
7	0	1	1	1
8	1	0	0	0
9	1	0	0	1
10	1	0	1	0
11	1	0	1	1
12	1	1	0	0
13	1	1	0	1
14	1	1	1	0
15	1	1	1	1

6.5 Secondary Amplitude Scale Factor Select

The Secondary Amplitude Scale Factor Select (ASF) is received from the Configuration Administrator via the SCCH. When transmitting the Hybrid or Extended Hybrid waveform, this field is ignored. When transmitting the All Digital waveform, changes to ASF can be effected seamlessly at an L1 frame boundary without discontinuity or disruption in Layer 1 service.

The upper and lower secondary sidebands may be independently scaled in amplitude. One of the four secondary amplitude scale factors, a_{2U} through a_{5U}, are selected by the user for application to the upper secondary sidebands. Likewise, one of the four secondary amplitude scale factors, a_{2L} through a_{5L}, are selected by the user for application to the lower secondary sidebands.

Table 6-7 defines the ASF bit mapping.

Table 6-7: Secondary Amplitude Scale Factor (ASF) Bit Mapping

Lower Sideband ASF	Upper Sideband ASF	ASF Bit Assignment (bits 3:0)			
		3	2	1	0
a_{2L}	a_{2U}	0	0	0	0
a_{3L}	a_{2U}	0	0	0	1
a_{4L}	a_{2U}	0	0	1	0
a_{5L}	a_{2U}	0	0	1	1
a_{2L}	a_{3U}	0	1	0	0
a_{3L}	a_{3U}	0	1	0	1
a_{4L}	a_{3U}	0	1	1	0
a_{5L}	a_{3U}	0	1	1	1
a_{2L}	a_{4U}	1	0	0	0
a_{3L}	a_{4U}	1	0	0	1
a_{4L}	a_{4U}	1	0	1	0
a_{5L}	a_{4U}	1	0	1	1
a_{2L}	a_{5U}	1	1	0	0
a_{3L}	a_{5U}	1	1	0	1
a_{4L}	a_{5U}	1	1	1	0
a_{5L}	a_{5U}	1	1	1	1

6.6 Primary Amplitude Scale Factors

The primary sideband scale factors (a_{0L}, a_{0U}, a_{1L}, and a_{1U}) are used to set the primary sideband power depending on the selected service mode. These values are received from the Configuration Administrator via the SCCH. The power level of the upper and lower sidebands may be set independently. Refer to [7] for further details.

The update rate and resolution of the Primary Amplitude Scale Factors, and the actual point or points within the signal path where they are applied depends on the specific implementation.

6.7 Reserved Control Data

The primary system control data sequence contains two bits that are designated reserved and the secondary system control data sequence contains eight bits that are designated reserved. These bits are controlled by the Configuration Administrator. The assignment of these bits to positions in the system control data sequence is specified in Table 6-8 and Table 6-9.

Table 6-8: Correlation of Primary Reserved Control Data Bits and System Control Data Sequence Bit Locations

Primary Reserved Control Data Bit #	**Primary System Control Data Sequence Bit #**
0	Primary system control data sequence bit 7
1	Primary system control data sequence bit 16

Table 6-9: Correlation of Secondary Reserved Control Data Bits and System Control Data Sequence Bit Locations

Secondary Reserved Control Data Bit #	**Secondary System Control Data Sequence Bit #**
0	Secondary system control data sequence bit 4
1	Secondary system control data sequence bit 5
2	Secondary system control data sequence bit 6
3	Secondary system control data sequence bit 7
4	Secondary system control data sequence bit 8
5	Secondary system control data sequence bit 16
6	Secondary system control data sequence bit 19
7	Secondary system control data sequence bit 24

7 Logical Channels

7.1 Introduction

A logical channel is a signal path that conducts L2 PDUs through Layer 1 with a specified grade of service. The primary logical channels are P1, P2, P3, P4, POV, PIDS, and PIDSOV. The secondary logical channels are S1 and SIDS. Logical channels are defined by their characterization parameters and configured by the service mode.

7.2 Characterization Parameters

For a given service mode, the grade of service of a particular logical channel may be uniquely quantified using three characterization parameters: transfer, latency, and robustness. Modulation, channel code rate, interleaver depth, digital diversity delay, and spectral mapping are the determinants of the characterization parameters.

7.2.1 Transfer

Transfer defines the throughput of a logical channel. The block-oriented operations of Layer 1 (such as interleaving) require that it process data in discrete transfer frames rather than continuous streams. As a result, throughput is defined in terms of *transfer frame size* (in bits) and *transfer frame rate* (in Hz, or the number of transfer frames per second). This Layer 1 framing effectively defines the alignment of L2 PDUs.

Each transfer frame is uniquely identified by its *transfer frame number*. The notation for the transfer frame number is presented as follows:

$$F^{n}_{m1:m2}$$

In the notation, the superscript n is the ALFN with which the transfer frame is associated and the subscript m1:m2 is the *BC range* that is spanned by the transfer frame within L1 frame n. Thus, the BC range indicates the position of the transfer frame within the L1 frame. The transfer frame number is not broadcast as part of the transmitted HD Radio signal.

All transfer frames are conducted through Layer 1 at one of three rates:

- the *L1 frame rate*, $R_f = \frac{1}{T_f}$
- the *L1 block rate*, $R_b = \frac{1}{T_b}$
- the *L1 block pair rate*, $R_p = \frac{1}{T_p}$

The ratio of the transfer frame rate to the L1 frame rate is termed the *transfer frame modulus*. For a transfer frame modulus of 1, the BC range is always 0:15. For a transfer frame modulus of 16, the BC range is always a single integer between 0 and 15. The transfer frame rate relationships are summarized in Table 7-1 and the transfer frame number timing relationships are illustrated in Figure 7-1.

Table 7-1: Transfer Frame Rate Relationships

Transfer Frame Type	Transfer Frame Modulus	Transfer Frame Duration (seconds)	Transfer Frame Rate (Hz)
L1 Block	16	T_b	$R_b = 16 \cdot R_f$
L1 Block Pair	8	$T_p = 2 \cdot T_b$	$R_p = 8 \cdot R_f$
L1 Frame	1	$T_f = 16 \cdot T_b$	R_f

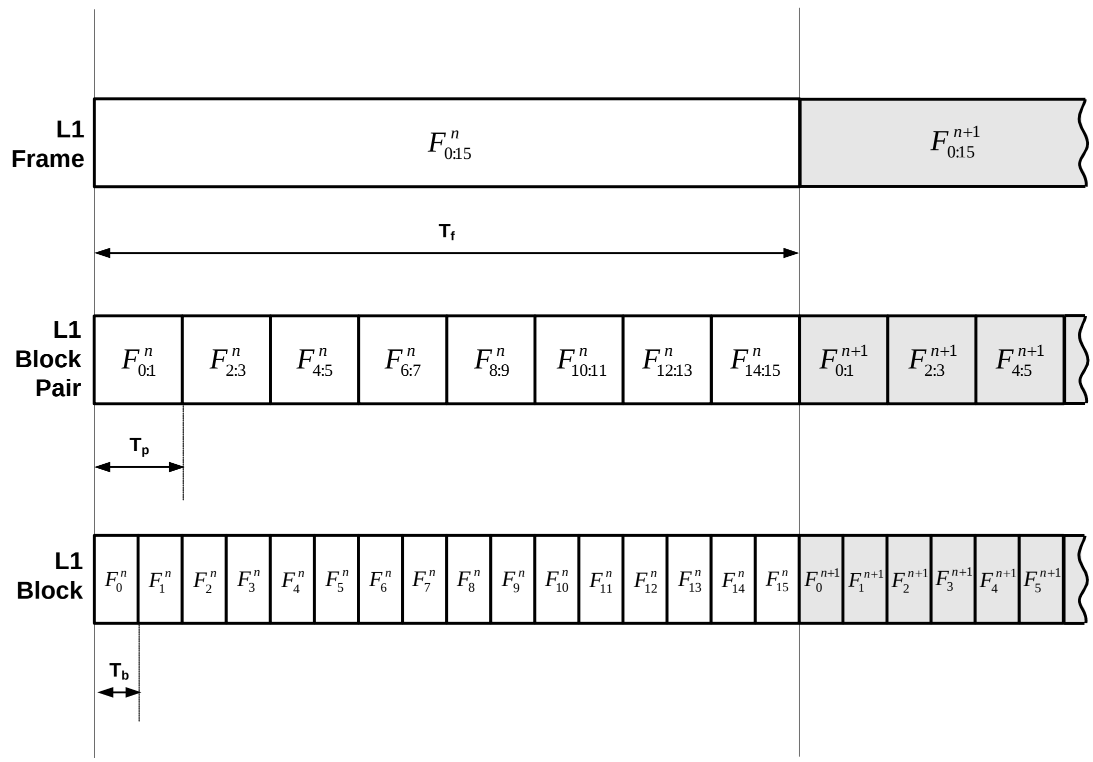


Figure 7-1: Transfer Frame Number Timing Relationship

Spectral mapping, channel code rate, and modulation determine the transfer of a logical channel since spectral mapping limits capacity, coding overhead limits information throughput, and modulation order determines the number of code bits that can modulate each OFDM subcarrier. Interleaver depth is also a factor because transfer frames are normally conducted through Layer 1 at rates corresponding to the interleaver depth of their logical channel.

7.2.2 Latency

Latency is the delay that a logical channel imposes on a transfer frame as it traverses Layer 1. The latency of a logical channel is defined as the sum of its interleaver depth and digital diversity delay. It does not include processing delays in Layer 1 nor does it include delays imposed in upper layers.

The interleaver depth determines the amount of delay imposed on a logical channel by an interleaver. The FM HD Radio system employs four interleaver depths: L1 block, *L1 block pair*, L1 frame, and *L1 frame pair*. A digital diversity delay of T_{dd} is also employed on some logical channels. For example, in some service modes, logical channel P1 presents dual processing paths; one path is delayed by T_{dd} from the other at the transmitter. In advanced service modes, some logical channels are split into main and backup components, with the backup component delayed by T_{dd} from the main component.

Higher layers assign information to logical channels with the requisite latency through service mode selection. Six latencies are specified for the system as defined in Table 7-2.

Table 7-2: Latency Summary

Description	Delay
L1 Block	T_b
L1 Block Pair	T_p
L1 Frame	T_f
L1 Frame Pair	$2 \cdot T_f$
L1 Block Pair plus Digital Diversity Delay	$T_p + T_{dd}$
L1 Frame plus Digital Diversity Delay	$T_f + T_{dd}$

7.2.3 Robustness

Robustness is the ability of a logical channel to withstand channel impairments such as noise, interference, and fading. There are nine relative levels of robustness in Layer 1 of the FM air interface. A robustness of 1 indicates a very high level of resistance to channel impairments while a robustness of 9 indicates a lower tolerance for channel-induced errors. As with latency, higher layers must determine the required robustness of a logical channel before selecting a service mode.

Modulation order, spectral mapping, channel code rate, interleaver depth, the power level of each sideband, and digital diversity delay determine the robustness of a logical channel. Modulation order affects robustness by fixing the relative distance among digital constellation points. Spectral mapping affects robustness by setting the relative power level, spectral interference protection, and frequency diversity of a logical channel. Channel coding increases robustness by introducing redundancy into the logical channel. Interleaver depth influences performance in multipath fading and brief signal outages, thereby affecting the robustness of the logical channel. Increasing the power of one or both primary or secondary sidebands will have a commensurate impact on robustness. Finally, some logical channels in certain service modes delay transfer frame components by a fixed duration to realize time diversity. This digital diversity delay also affects robustness since it mitigates the effects of the mobile radio channel.

7.2.4 Assignment of Characterization Parameters

Table 7-3 through Table 7-16 show the characterization parameters of each logical channel for each service mode. Transfer is presented in terms of transfer frame size, transfer frame rate, and transfer frame modulus. The relative robustness figures are approximate. Exact performance may vary depending on the specific channel conditions as well as individual sideband power. The robustness figures provided in the following tables assume equal power levels in each primary sideband. Secondary logical channels are assigned the lowest robustness level, since their power will likely be significantly lower than the level of the primary sidebands.

Table 7-3: Logical Channel Characterization – Service Mode MP1

Logical Channel	Transfer			Latency (seconds)	Relative Robustness
	Frame Size (bits)	Frame Rate (Hz)	Frame Modulus		
P1	146176	R_f	1	T_f	2
PIDS	80	R_b	16	T_b	3

Table 7-4: Logical Channel Characterization – Service Mode MP2

Logical Channel	Transfer			Latency (seconds)	Relative Robustness
	Frame Size (bits)	Frame Rate (Hz)	Frame Modulus		
P1	146176	R_f	1	T_f	2
P3	2304	R_p	8	$2 \cdot T_f$	2
PIDS	80	R_b	16	T_b	3

Table 7-5: Logical Channel Characterization – Service Mode MP3

Logical Channel	Transfer			Latency (seconds)	Relative Robustness
	Frame Size (bits)	Frame Rate (Hz)	Frame Modulus		
P1	146176	R_f	1	T_f	2
P3	4608	R_p	8	$2 \cdot T_f$	2
PIDS	80	R_b	16	T_b	3

Table 7-6: Logical Channel Characterization – Service Mode MP11

Logical Channel	Transfer			Latency (seconds)	Relative Robustness
	Frame Size (bits)	Frame Rate (Hz)	Frame Modulus		
P1	146176	R_f	1	T_f	2
P3	4608	R_p	8	$2 \cdot T_f$	2
P4	4608	R_p	8	$2 \cdot T_f$	2
PIDS	80	R_b	16	T_b	3

Table 7-7: Logical Channel Characterization – Service Mode MP5

Logical Channel	Transfer			Latency (seconds)	Relative Robustness
	Frame Size (bits)	Frame Rate (Hz)	Frame Modulus		
P1	4608	R_p	8	$T_p + T_{dd}$	1
P2	109312	R_f	1	T_f	2
P3	4608	R_p	8	$2 \cdot T_f$	2
PIDS	80	R_b	16	T_b	3

Table 7-8: Logical Channel Characterization – Service Mode MP6

Logical Channel	Transfer			Latency (seconds)	Relative Robustness
	Frame Size (bits)	Frame Rate (Hz)	Frame Modulus		
P1	9216	R_p	8	$T_p + T_{dd}$	1
P2	72448	R_f	1	T_f	2
PIDS	80	R_b	16	T_b	3

Table 7-9: Logical Channel Characterization – Service Mode MP1X

Logical Channel	Transfer			Latency (seconds)	Relative Robustness
	Frame Size (bits)	Frame Rate (Hz)	Frame Modulus		
P1	146176	R_f	1	T_f	2
P4	13824	R_p	8	$T_p + T_{dd}$	2
PIDS	80	R_b	16	T_b	3

Table 7-10: Logical Channel Characterization – Service Mode DSB1

Logical Channel	Transfer			Latency (seconds)	Relative Robustness
	Frame Size (bits)	Frame Rate (Hz)	Frame Modulus		
P1	45440	R_p	8	$T_p + T_{dd}$	2
PIDS	320	R_b	16	T_b	4

Table 7-11: Reserved

Table 7-12: Logical Channel Characterization – Service Mode MP1XOV

Logical Channel	Transfer			Latency (seconds)	Relative Robustness
	Frame Size (bits)	Frame Rate (Hz)	Frame Modulus		
P1	146176	R_f	1	T_f	3
P4	6912	R_p	8	$T_p + T_{dd}$	1
PIDS	80	R_b	16	T_b	4
POV	22720	R_p	8	$T_p + T_{dd}$	2
PIDSOV	160	R_b	16	T_b	5

Table 7-13: Logical Channel Characterization – Service Mode MP6OV

Logical Channel	Transfer			Latency (seconds)	Relative Robustness
	Frame Size (bits)	Frame Rate (Hz)	Frame Modulus		
P1	9216	R_p	8	$T_p + T_{dd}$	1
P2	72448	R_f	1	T_f	3
PIDS	80	R_b	16	T_b	4
POV	22720	R_p	8	$T_p + T_{dd}$	2
PIDSOV	160	R_b	16	T_b	5

Table 7-14: Logical Channel Characterization – Service Mode DSB1OV

Logical Channel	Transfer			Latency (seconds)	Relative Robustness
	Frame Size (bits)	Frame Rate (Hz)	Frame Modulus		
P1	45440	R_p	8	$T_p + T_{dd}$	2
PIDS	320	R_b	16	T_b	5
POV	22720	R_p	8	$T_p + T_{dd}$	4
PIDSOV	160	R_b	16	T_b	6

Table 7-15: Reserved

Table 7-16: Logical Channel Characterization – Service Mode MS5

Logical Channel	Transfer			Latency (seconds)	Relative Robustness
	Frame Size (bits)	Frame Rate (Hz)	Frame Modulus		
S1	22720	R_p	8	$T_p + T_{dd}$	8
SIDS	160	R_b	16	T_b	9

The throughput of a logical channel through Layer 1 can be calculated using these tables and the following formula:

$$throughput\,(bits/sec) = transfer\ frame\,size\,(bits) \cdot transfer\ frame\,rate\,(Hz)$$

For example, in service mode MP1, the throughput for logical channel P1 is calculated as follows:

$$throughput\,(bits/sec) = 146{,}176 \cdot \frac{44{,}100}{65{,}536} \approx 98.4\ kbit/sec$$

Note that the throughput represents the rate of data transfer through Layer 1, and not necessarily the information bit rate. This is because some logical channels in an L2 PDU may contain a small amount of overhead (e.g., framing, parity, etc.) that is appended in the upper protocol layers to the audio or data content.

7.3 Logical Channel Spectral Mapping

For a given service mode, each logical channel is applied to a group of OFDM subcarriers or frequency partitions as illustrated in Figure 7-2 through Figure 7-15. In these figures, the annotated frequencies represent offsets from the channel center frequency.

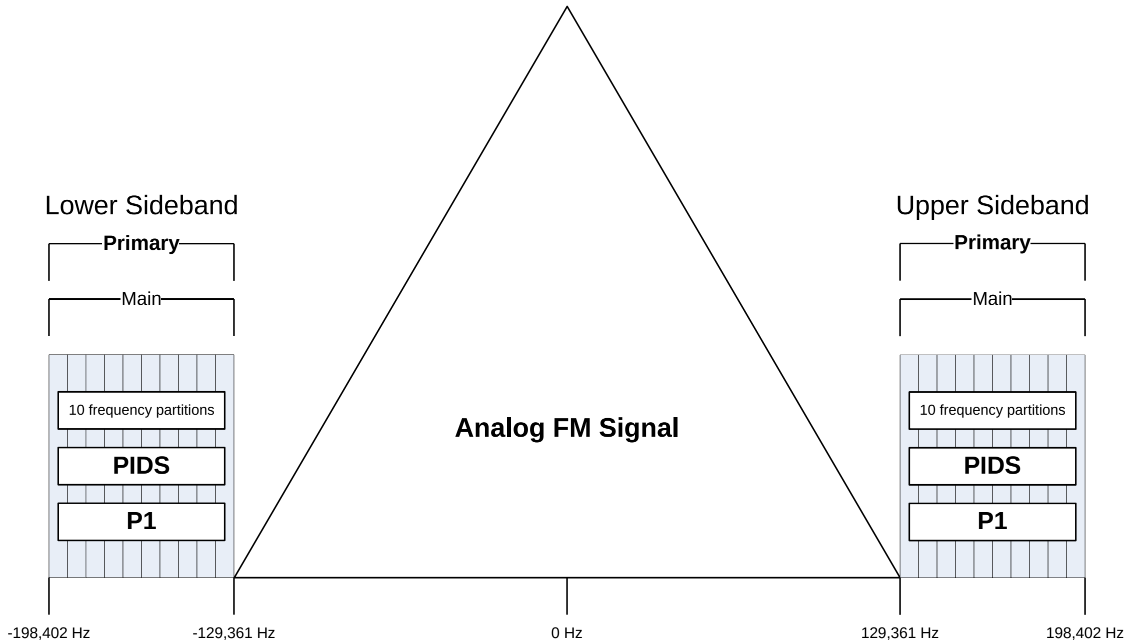


Figure 7-2: Logical Channel Spectral Mapping – Service Mode MP1

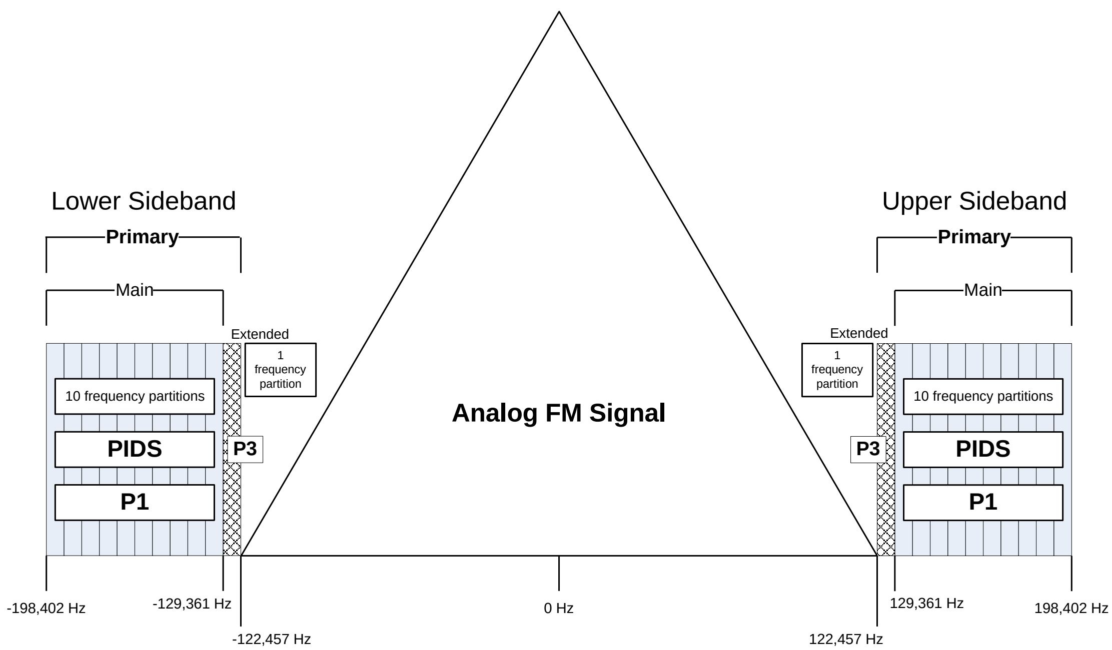


Figure 7-3: Logical Channel Spectral Mapping – Service Mode MP2

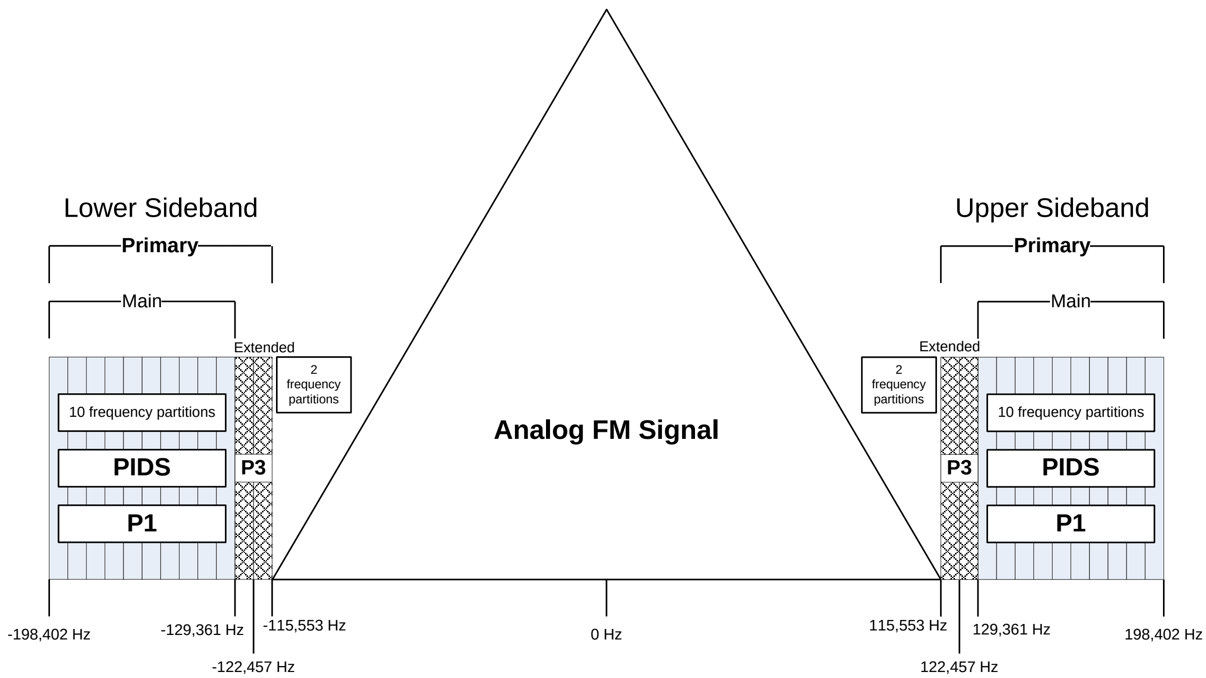


Figure 7-4: Logical Channel Spectral Mapping – Service Mode MP3

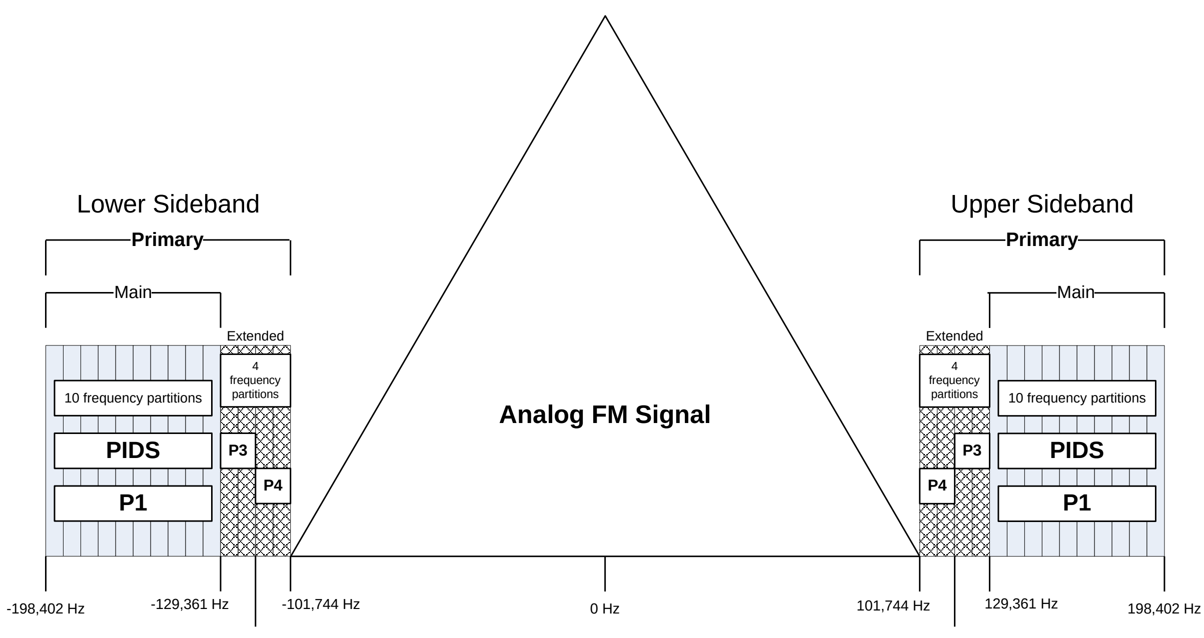


Figure 7-5: Logical Channel Spectral Mapping – Service Mode MP11

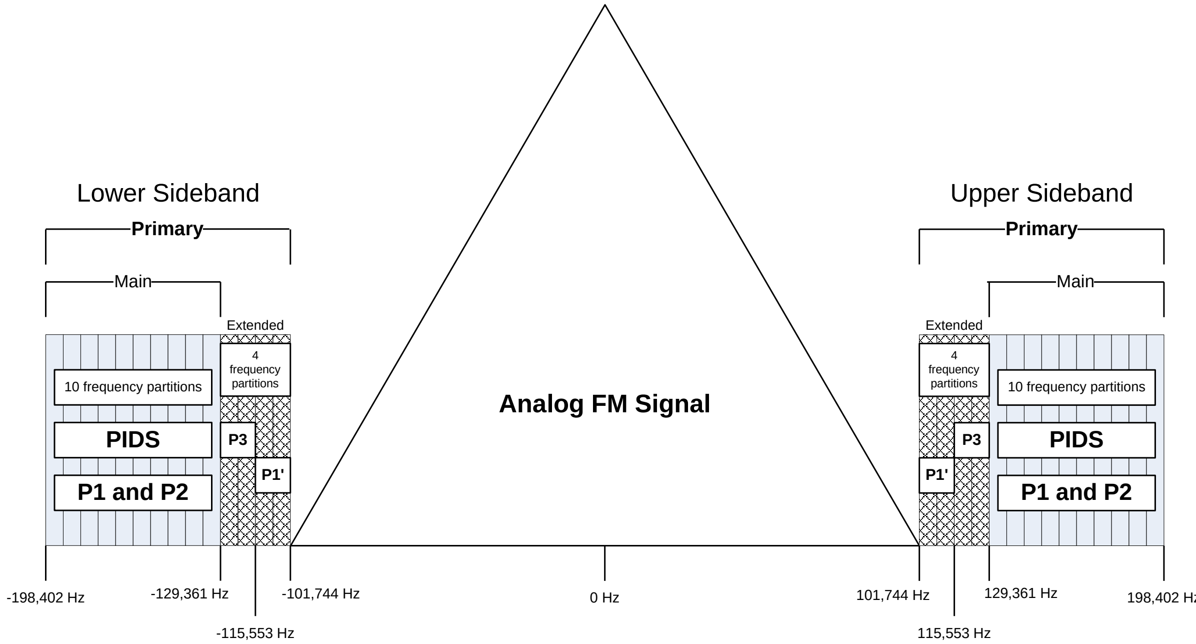


Figure 7-6: Logical Channel Spectral Mapping – Service Mode MP5

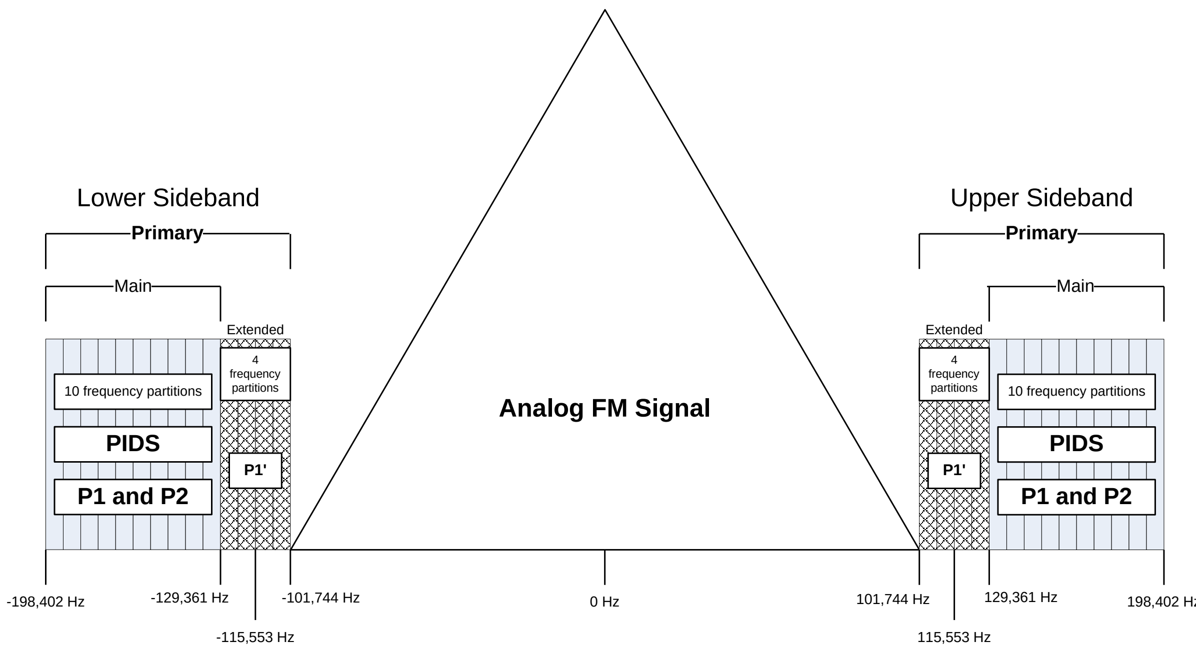


Figure 7-7: Logical Channel Spectral Mapping – Service Mode MP6

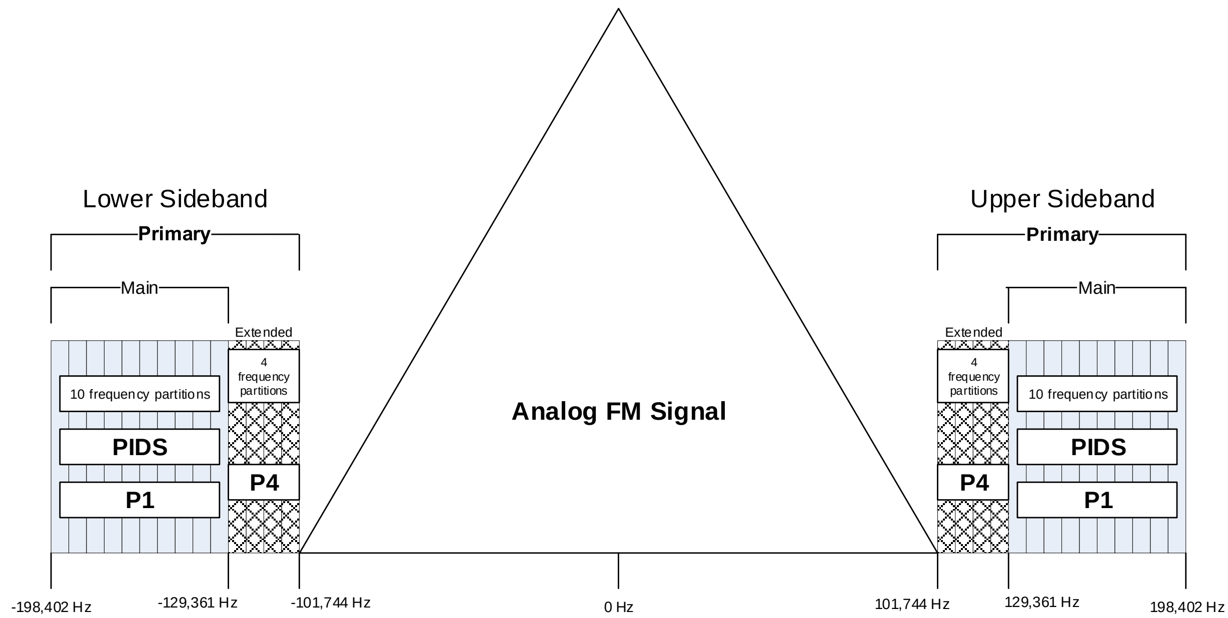


Figure 7-8: Logical Channel Spectral Mapping – Service Mode MP1X

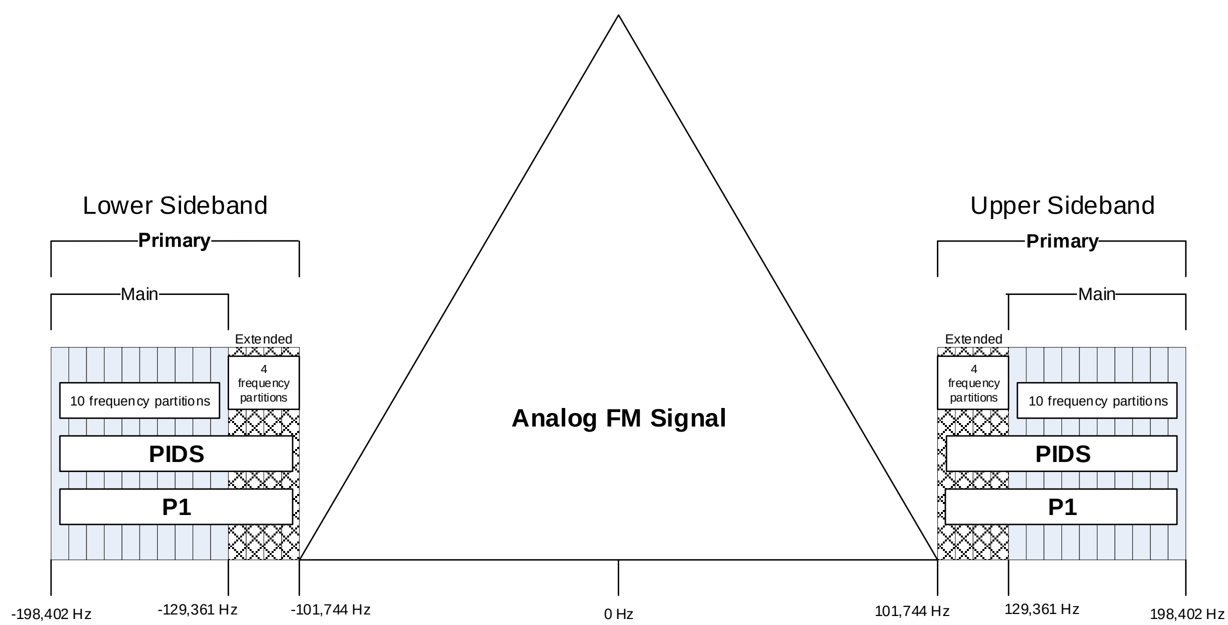


Figure 7-9: Logical Channel Spectral Mapping – Service Mode DSB1

Figure 7-10: Reserved

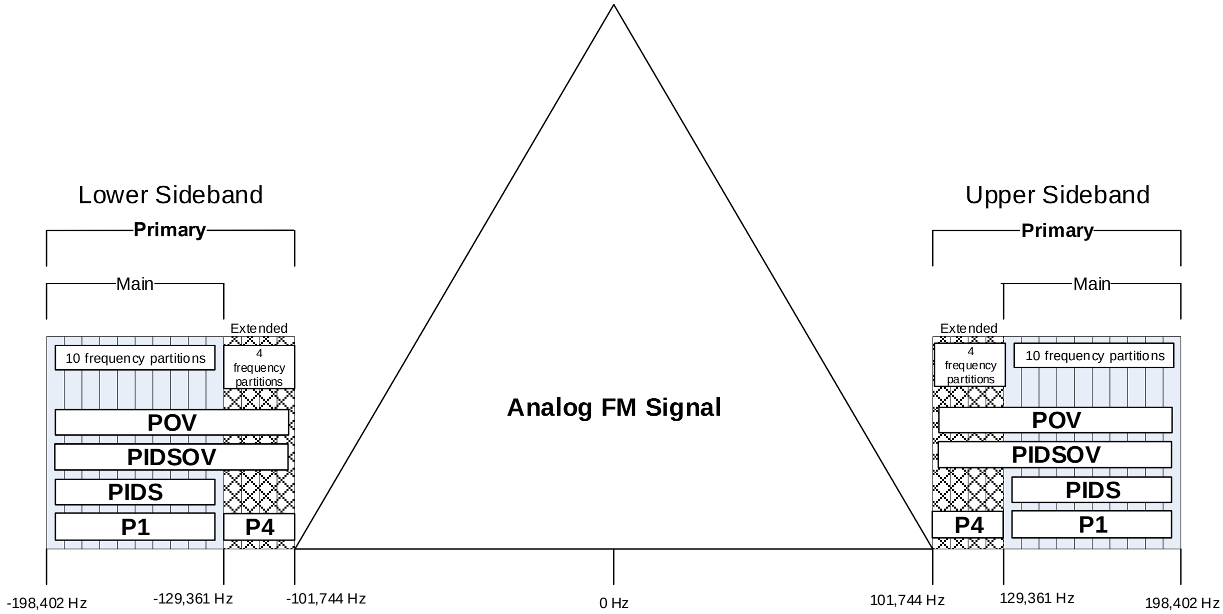


Figure 7-11: Logical Channel Spectral Mapping – Service Mode MP1XOV

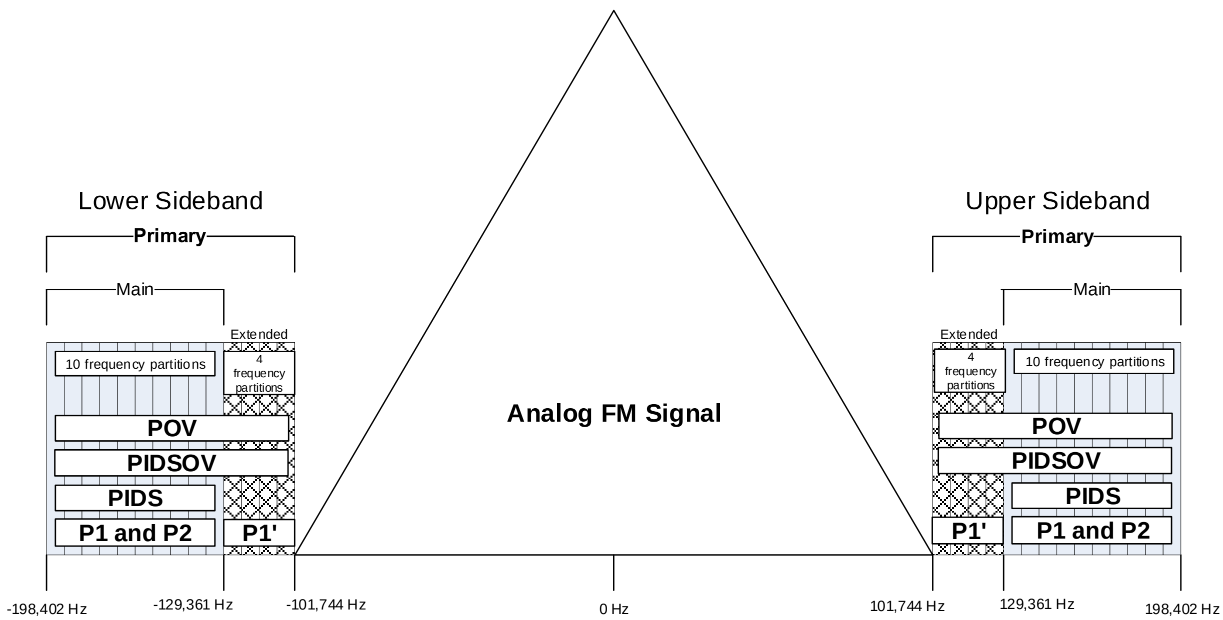


Figure 7-12: Logical Channel Spectral Mapping – Service Mode MP6OV

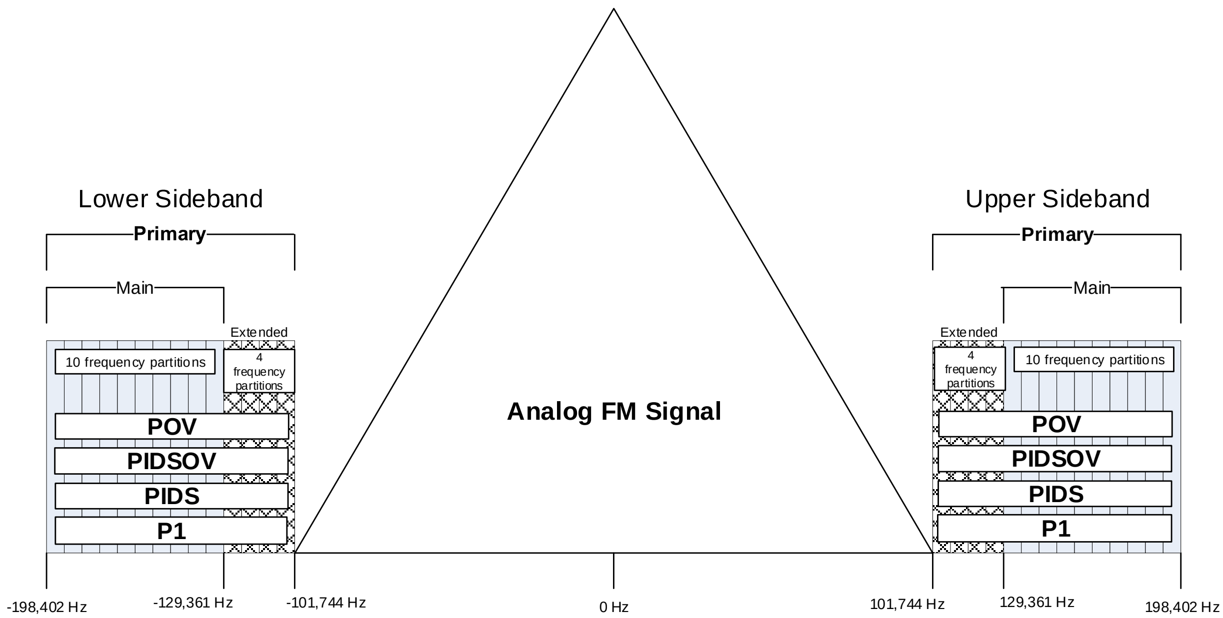


Figure 7-13: Logical Channel Spectral Mapping – Service Mode DSB1OV

Figure 7-14: Reserved

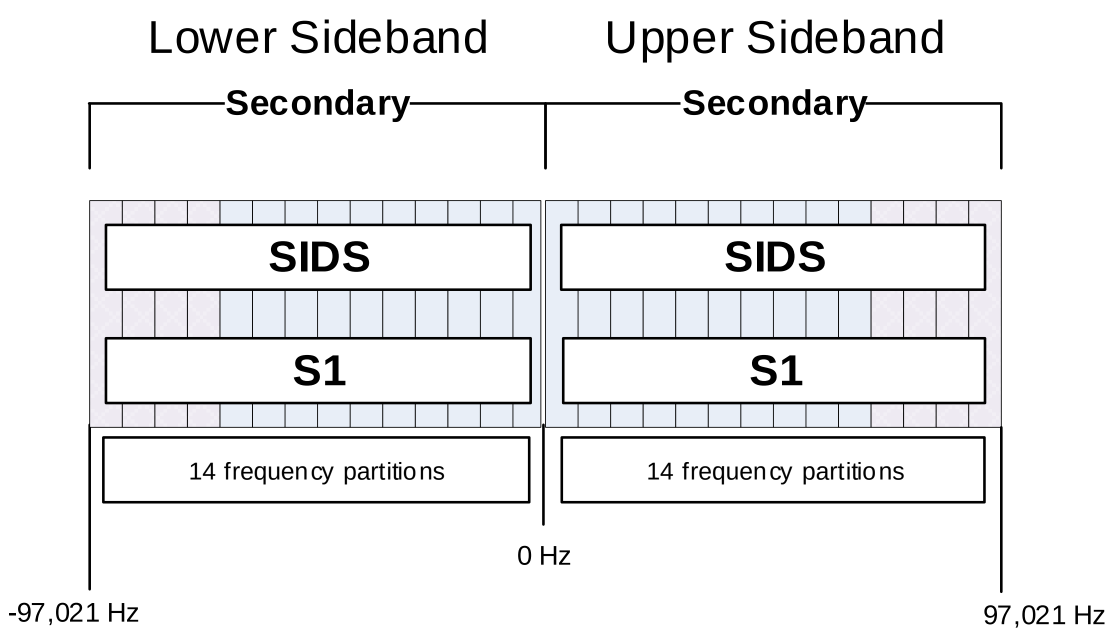


Figure 7-15: Logical Channel Spectral Mapping – Service Mode MS5

7.4 Logical Channel Framing and Synchronization

The logical channels share a common, absolute time reference so that all transfer frames are precisely aligned. As described in Subsection 7.2.1, each transfer frame is assigned a unique transfer frame number with the notation:

$$F^{n}_{m1:m2}$$

where superscript *n* is the ALFN and subscript *m1:m2* is the BC range that designates the position of the transfer frame within the indexed L1 frame n.

This numbering scheme allows all transfer frames to be referenced to an absolute transmission time.

8 Scrambling

8.1 Introduction

The bits in each logical channel are scrambled to randomize the time-domain data and aid in receiver synchronization. As shown in Figure 8-1, there are nine parallel scramblers: one for each logical channel.

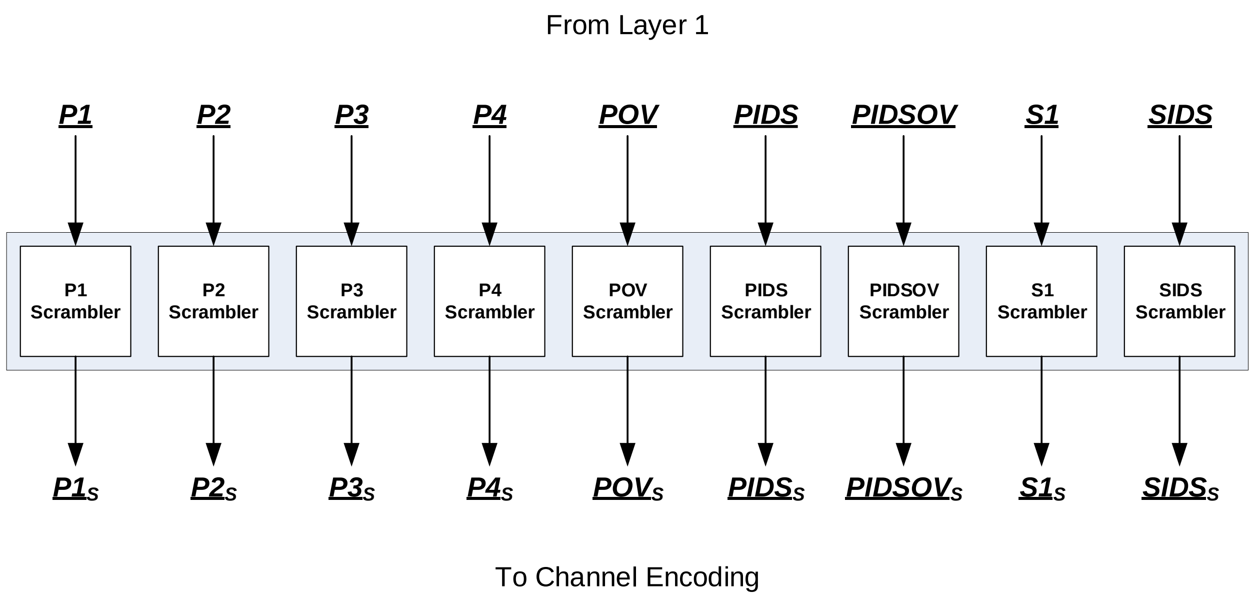


Figure 8-1: Scrambling Functional Block Diagram

The inputs to the scramblers are the active logical channels as selected by the service mode. These inputs are delivered in discrete transfer frames whose size and rate are defined in Table 7-3 through Table 7-16 for a given service mode. The outputs of the scramblers are transfer frames of scrambled bits for each of the active logical channels. These transfer frames are passed to the channel encoding process for forward error correction.

8.2 Scrambler Operation

All parallel scramblers are identical, but operate at different rates, depending on the active service mode. A detailed block diagram of the scrambler is shown in Figure 8-2. Each scrambler generates a maximal-length scrambling sequence using a linear feedback shift register with the following primitive polynomial:

$$P(x) = 1 \oplus x^2 \oplus x^{11}$$

A given bit of a scrambled transfer frame is generated by modulo-2 adding the associated input bit with the corresponding bit of the scrambling sequence.

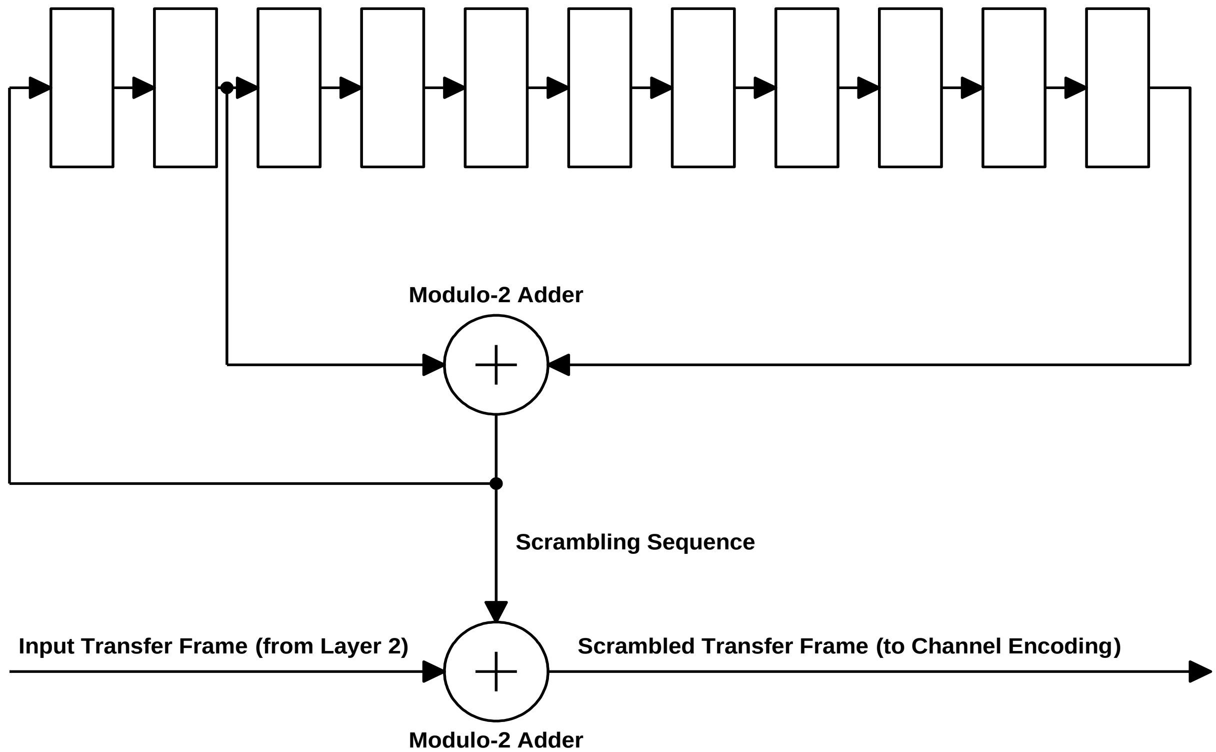


Figure 8-2: Scrambler Block Diagram

For each logical channel, the scrambler is reset to state 0111 1111 111 upon receipt of a new transfer frame. The first bit of a scrambled transfer frame is generated by modulo-2 adding the first bit of the input transfer frame with the scrambling bit generated when the shift register is set to the initial state. The process then continues until the last bit of the input transfer frame is scrambled.

9 Channel Encoding

9.1 Introduction

Channel encoding improves system performance by increasing the robustness of the signal in the presence of channel impairments. The channel encoding process is characterized by two main operations: time delay (for digital diversity delay and *transmit time alignment*) and convolutional encoding. The channel encoding process in shown in Figure 9-1 for standard primary service modes, Figure 9-2 for advanced primary service modes, and Figure 9-3 for secondary service mode MS5.

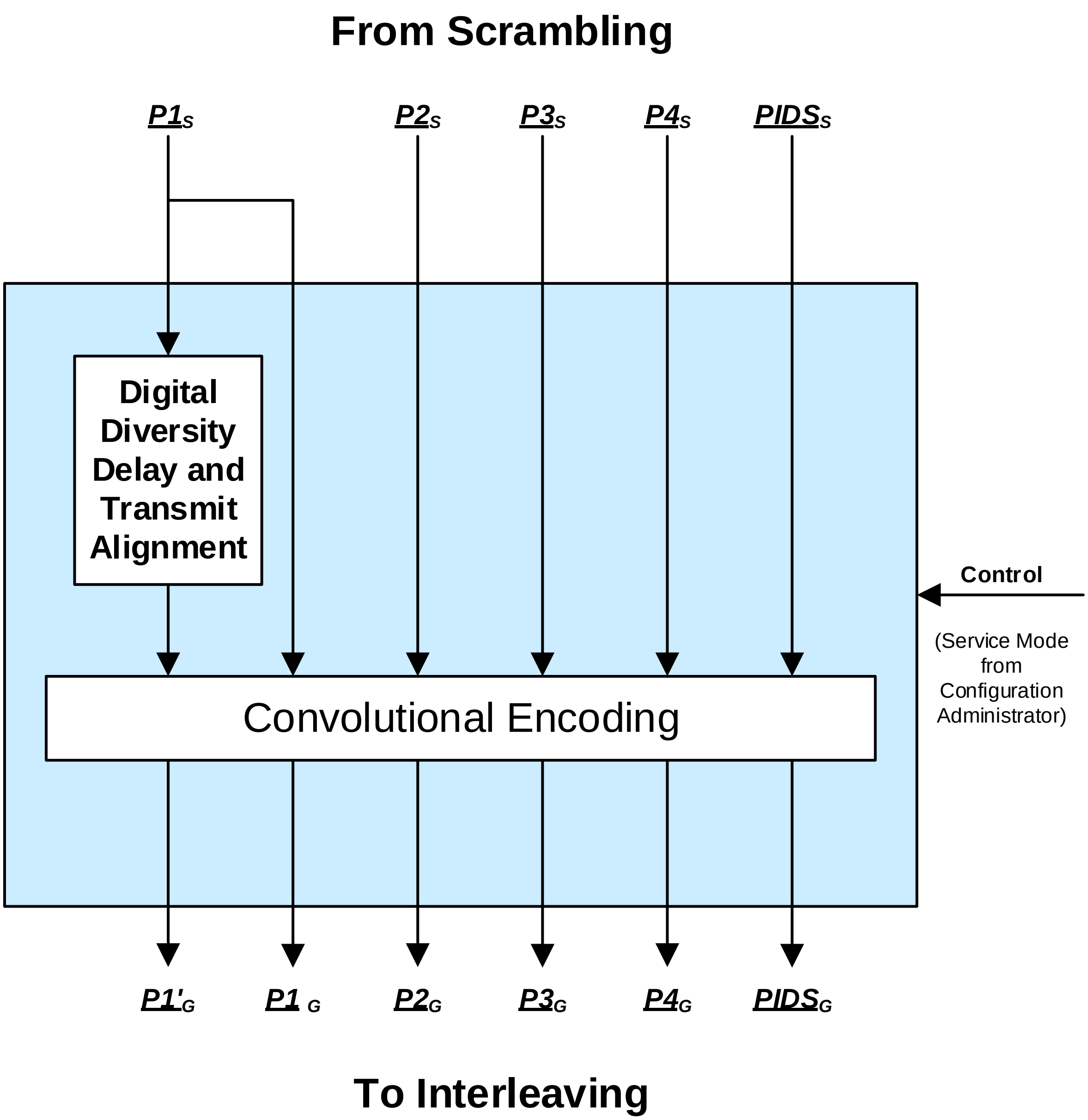


Figure 9-1: Channel Encoding Conceptual Block Diagram for Standard Primary Service Modes

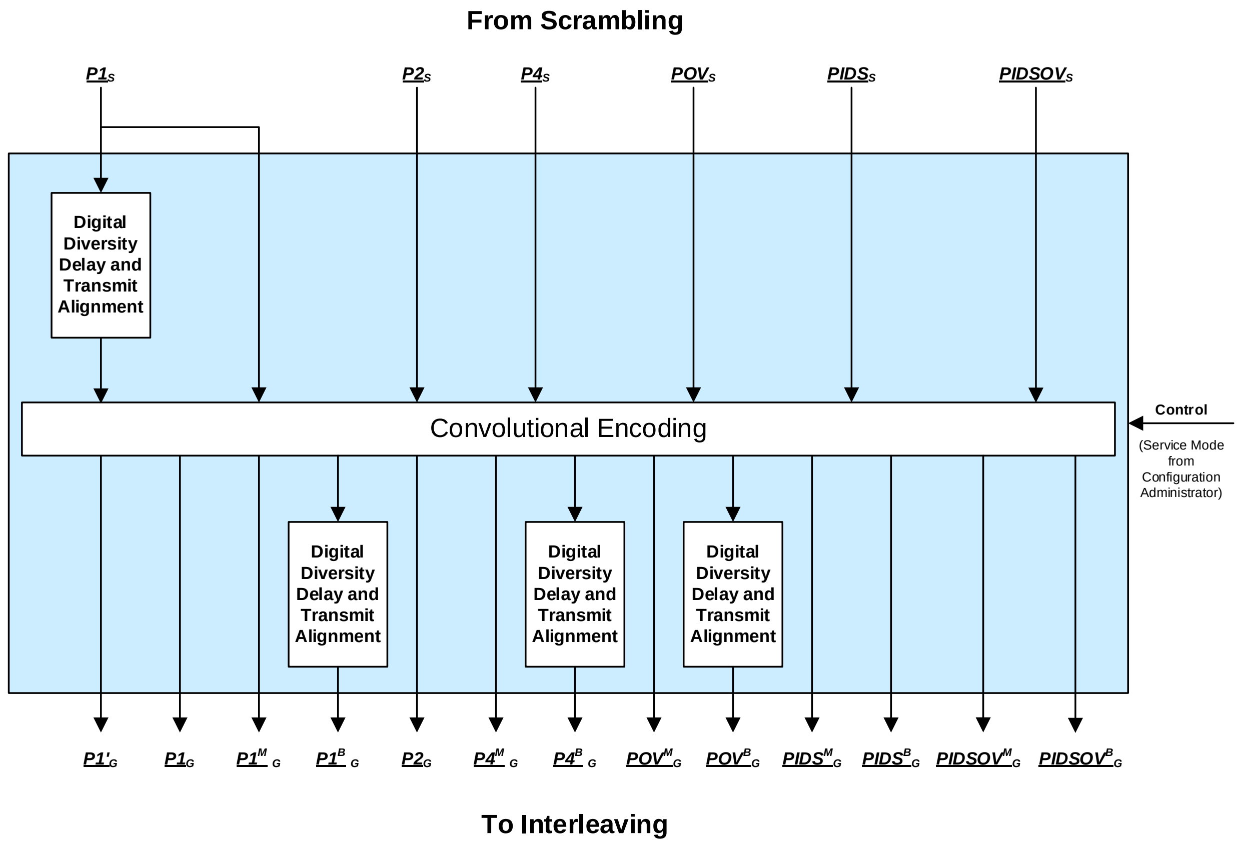


Figure 9-2: Channel Encoding Conceptual Block Diagram for Advanced Primary Service Modes

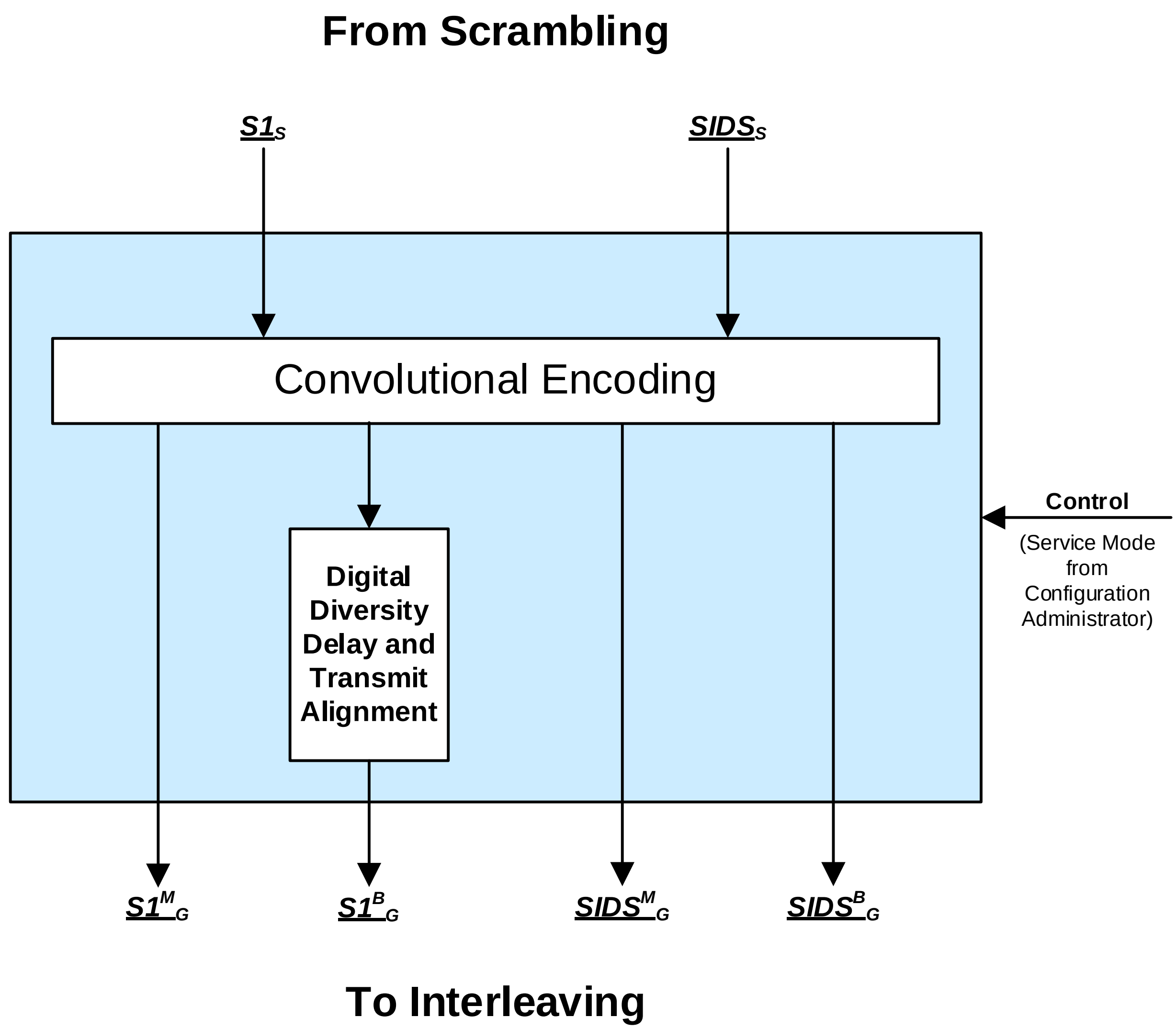


Figure 9-3: Channel Encoding Conceptual Block Diagram for Secondary Service Mode MS5

The inputs to the channel encoding process are transfer frames of scrambled bits carried through the active logical channels. The size and rate of transfer are defined in Table 7-3 through Figure 7-15 for a given service mode. The outputs of the channel encoding process are transfer frames of encoded bits associated with each of the active logical channels. The output transfer frames are passed to the interleaving function.

In the ensuing sections, for notational convenience, the logical channel vectors at a particular stage of processing are represented in shorthand notation by their subscript. For example, the scrambled inputs $\underline{Px}_S$ and $\underline{Sx}_S$ are represented by $\underline{S}$ while the encoded outputs $\underline{Px}_G$ and $\underline{Sx}_G$ are represented by $\underline{G}$. Also, the primed notation (as in $\underline{P1'}_G$) indicates that the logical channel vector is processed differently than the "unprimed" logical channel (for example see Figure 9-11, Figure 9-12, and Figure 9-16) and is destined for transmission in the primary extended sideband. Finally, some logical channels in advanced primary service modes and secondary service mode MS5 are divided by the convolutional encoder into main (denoted $\underline{Px}^M_G$ or $\underline{Sx}^M_G$) and backup (denoted $\underline{Px}^B_G$ or $\underline{Sx}^B_G$) components that are subject to different processing.

9.2 Digital Diversity Delay and Transmit Time Alignment

Depending on the service mode, logical channel P1 may be split into P1 and P1′ components, where P1′ is delayed as it enters the channel encoding process. Likewise, backup logical channels $P1^B$, $P4^B$, POV^B, and $S1^B$ in advanced service modes are delayed following the convolutional encoding. The manner in which digital diversity delay is applied to these logical channels is presented in Subsection 9.4, and illustrated in Figure 9-11 through Figure 9-18, for each service mode. The delay provides time diversity to the affected logical channels. If applied, the value of the digital diversity delay is fixed at $N_{dd} \cdot T_f$.

In cases where digital diversity delay is applied, an additional delay called transmit time alignment is imposed on the digital diversity-delayed signals. This alignment delay ensures that the delayed logical channels (P1′, $P1^B$, $P4^B$, POV^B, and $S1^B$) are precisely positioned in time relative to their un-delayed counterparts (P1, $P1^M$, $P4^M$, POV^M, and $S1^M$, respectively) with the same content to accommodate diversity combining in the receiver.

9.3 Convolutional Encoding

Convolutional encoding consists of three primary operations: *mother code* generation, *puncturing*, and parallel-to-serial conversion. Some logical channels in advanced service modes also require main/backup bit allocation after puncturing. Each of these operations is described below. A description of the codes employed in the FM system follows in Subsection 9.3.5.

9.3.1 Mother Code Generation

A convolutional encoder employs select generator polynomials to form a group of *mother codes*. A rate 1/n convolutional encoder outputs n encoded bits (symbolized in the matrix as $g_{h,i}$) for every input bit (s_i) of the scrambled input vector $\underline{S}$ creating a codeword matrix $\underline{\underline{G}}$ of dimension n x N:

$$\underline{\underline{G}} = \begin{bmatrix} g_{1,0} & g_{1,1} & \cdots & g_{1,N-1} \\ g_{2,0} & g_{2,1} & \cdots & g_{2,N-1} \\ \vdots & \vdots & \vdots & \vdots \\ g_{n,0} & g_{n,1} & \cdots & g_{n,N-1} \end{bmatrix}$$

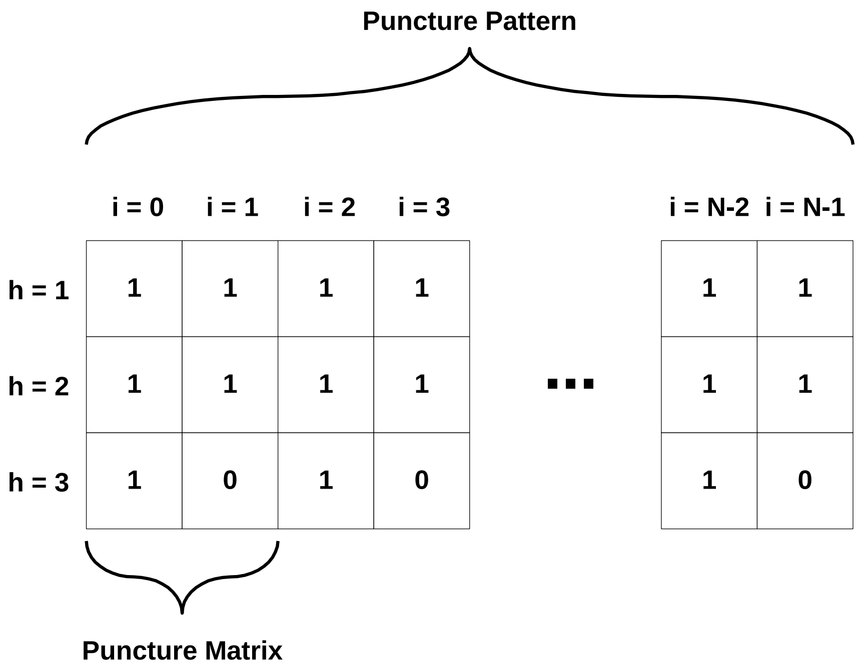
For the input bits s_i: $i = 0,1,2,\ldots,N-1$ where N is the length of $\underline{S}$.

h indexes the codeword bits for a given input bit and $h = 1,2,\ldots,n$. In the FM system, $n = 3$.

Each column of $\underline{\underline{G}}$ represents the encoded output for a given input bit.

9.3.2 Puncturing

Some service modes require puncturing of a *mother codeword* to produce a slightly higher code rate, thereby allowing a higher information rate through the same physical bandwidth. The codeword matrix $\underline{\underline{G}}$ is punctured over a puncture period P. For every P encoded bits, certain bits ($g_{h,i}$) are not transmitted. A puncture matrix spanning the encoded bits over a puncture period defines which encoded bits are transmitted. Repeating the puncture matrix over all encoded bits of a transfer frame forms the puncture pattern.


Puncture Pattern
i = 0
i = 1
i = 2
i = 3
i = N-2
i = N-1
h = 1
h = 2
h = 3
1 1 1 1 1 1
1 1 1 1 1 1
1 0 1 0 1 0
Puncture Matrix

9.3.3 Main/Backup Bit Allocation

In addition to puncturing, some logical channels in advanced service modes distribute the encoded bits between main and backup components. Each component has a code rate that is twice the rate of the composite code comprised of both main and backup components.

9.3.4 Parallel-to-Serial Conversion

After the mother code bits are appropriately punctured, the parallel-to-serial converter multiplexes them by concatenating the columns of $\underline{\underline{G}}$ into a single vector $\underline{G}$ as follows:

$$\underline{G} = [g_{1,0}, g_{2,0}, \ldots, g_{n,0}, g_{1,1}, g_{2,1}, \ldots, g_{n,1}, \ldots, g_{1,N-1}, g_{2,N-1}, \ldots, g_{n,N-1}]$$

For logical channels in advanced service modes that distribute encoded bits between main and backup components, vectors $\underline{BG}$ and $\underline{MG}$ are produced with constituent encoded bits defined by the puncture pattern.

9.3.5 Convolutional Encoders

Table 9-1 presents the code rates used in the FM system along with their associated puncture matrices, main/backup bit allocations, and constraint lengths, K. All codes use a mother code of rate 1/3. Punctured bits that are not transmitted are denoted with a 0 in the puncture matrix.

For those logical channels in advanced service modes that are split into main and backup components, encoded bits directed to the main component are denoted by M, and encoded bits directed to the backup component are denoted with a B.

Table 9-1: FM Convolutional Codes

Punctured Code Rate	Puncture Matrix	Main/Backup Bit Allocation	Constraint Length, K
2/5	$\begin{bmatrix} 1 & 1 \\ 1 & 1 \\ 1 & 0 \end{bmatrix}$	N/A	7
1/2	$\begin{bmatrix} 1 & 1 \\ 0 & 0 \\ 1 & 1 \end{bmatrix}$	N/A	7
3/8	$\begin{bmatrix} 1 & 1 & 1 & 1 & 1 & 1 \\ 1 & 1 & 1 & 1 & 1 & 1 \\ 1 & 1 & 0 & 0 & 1 & 1 \end{bmatrix}$	$\begin{bmatrix} 0 & M & 0 & M & 0 & 0 \\ 0 & 0 & M & 0 & M & M \\ M & 0 & 0 & 0 & M & M \end{bmatrix}$ $\begin{bmatrix} B & 0 & B & 0 & B & B \\ B & B & 0 & B & 0 & 0 \\ 0 & B & 0 & 0 & 0 & 0 \end{bmatrix}$	9
5/14	$\begin{bmatrix} 1 & 1 & 1 & 1 & 0 \\ 1 & 1 & 1 & 1 & 1 \\ 1 & 1 & 1 & 1 & 1 \end{bmatrix}$	$\begin{bmatrix} 0 & 0 & 0 & 0 & 0 \\ M & M & 0 & M & M \\ 0 & M & M & 0 & M \end{bmatrix}$ $\begin{bmatrix} B & B & B & B & 0 \\ 0 & 0 & B & 0 & 0 \\ B & 0 & 0 & B & 0 \end{bmatrix}$	9

A detailed description of each of these codes is provided in Subsection 9.3.5.2 through Subsection 9.3.5.5. The last K-1 bits of a given transfer frame are used to initialize the delay elements of the corresponding convolutional encoder for that transfer frame. The initial state of the encoder is set to $\underline{S}$[N-(K-1)], $\underline{S}$[N-(K-2)], …, $\underline{S}$ [N-1] prior to inputting $\underline{S}$[0] for every input transfer frame period. $\underline{S}$[N-(K-1)] represents the rightmost delay element and $\underline{S}$[N-1] represents the leftmost delay element shown in the figures. The fact that transfer frames define the encoding blocks is important in maintaining alignment between different logical channels.

9.3.5.1 Rate 1/3 Code

The rate 1/3 mother codes for constraint lengths K=7 and K=9 are defined by the generator polynomials shown in Table 9-2 (represented in octal format).

Table 9-2: Convolutional Encoder Generator Polynomials – Rate 1/3 Mother Code

Constraint Length, K	First Generator	Second Generator	Third Generator
7	133	171	165
9	557	663	711

9.3.5.2 Rate 2/5 Punctured Code

The K=7 rate 1/3 mother code is punctured to produce a rate 2/5 code. The rate 2/5 convolutional encoder is illustrated in Figure 9-4.

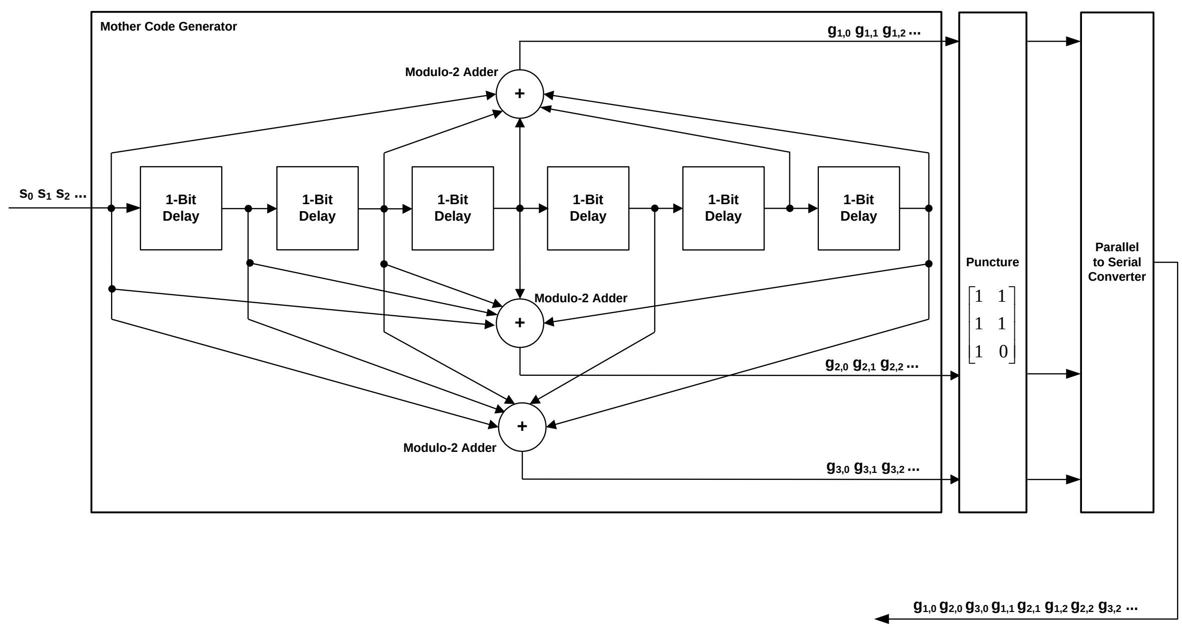


Figure 9-4: Convolutional Encoder – Rate 2/5 Code

9.3.5.3 Rate 1/2 Punctured Code

The K=7 rate 1/3 mother code is also punctured to produce a rate 1/2 code. The rate 1/2 convolutional encoder is illustrated in Figure 9-5.

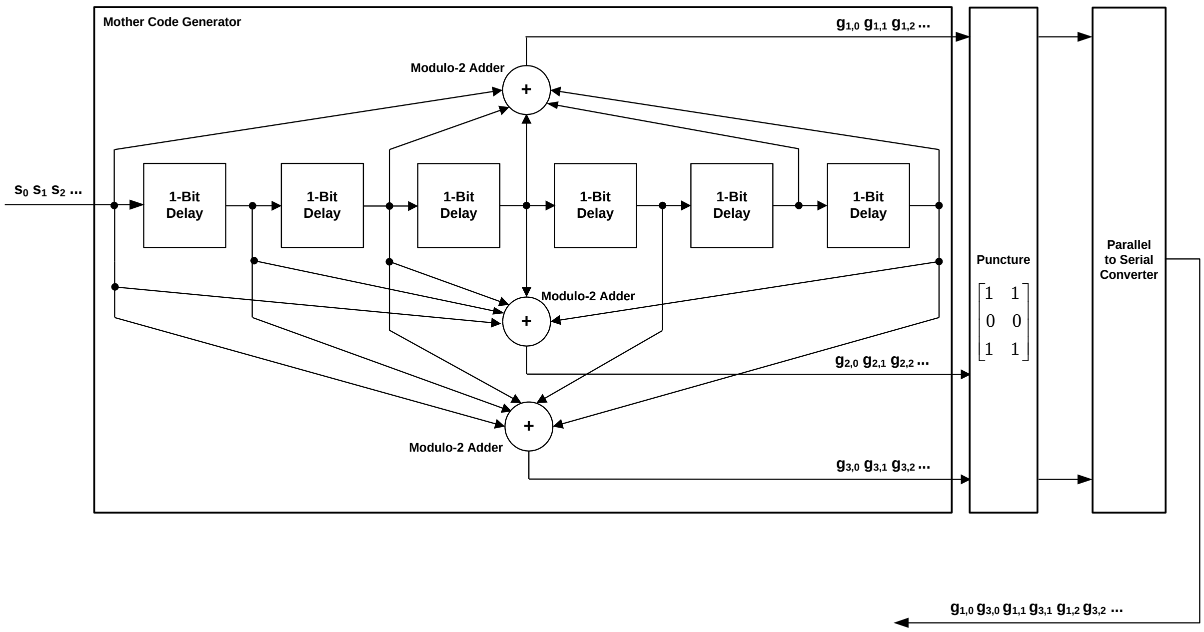


Figure 9-5: Convolutional Encoder – Rate 1/2 Code

9.3.5.4 Rate 3/8 Punctured Code

The K=9 rate 1/3 mother code is punctured to produce a rate 3/8 code with main and backup components. The rate 3/8 convolutional encoder is illustrated in Figure 9-6.

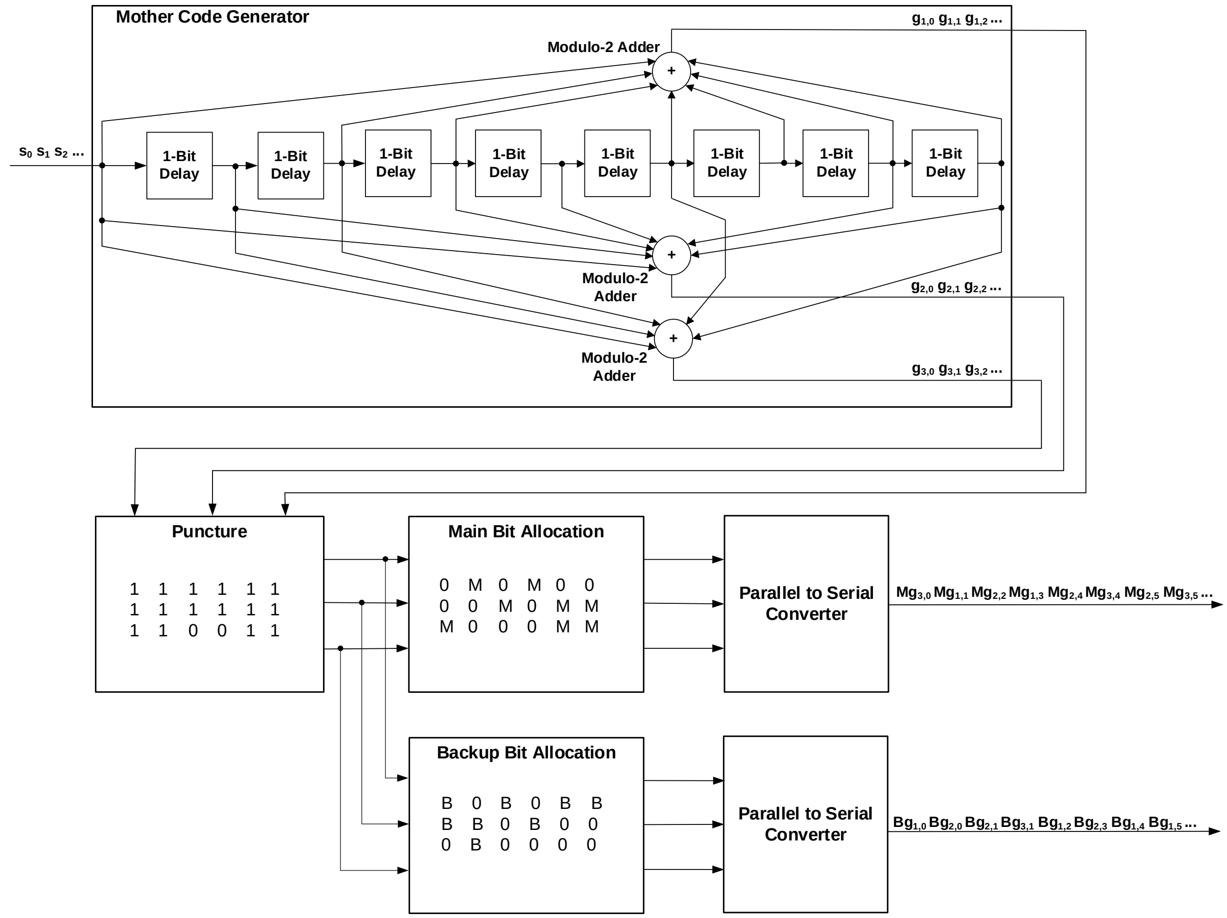


Figure 9-6: Convolutional Encoder – Rate 3/8 Code

9.3.5.5 Rate 5/14 Punctured Code

The K=9 rate 1/3 mother code is also punctured to produce a rate 5/14 code with main and backup components. The rate 5/14 convolutional encoder is illustrated in Figure 9-7.

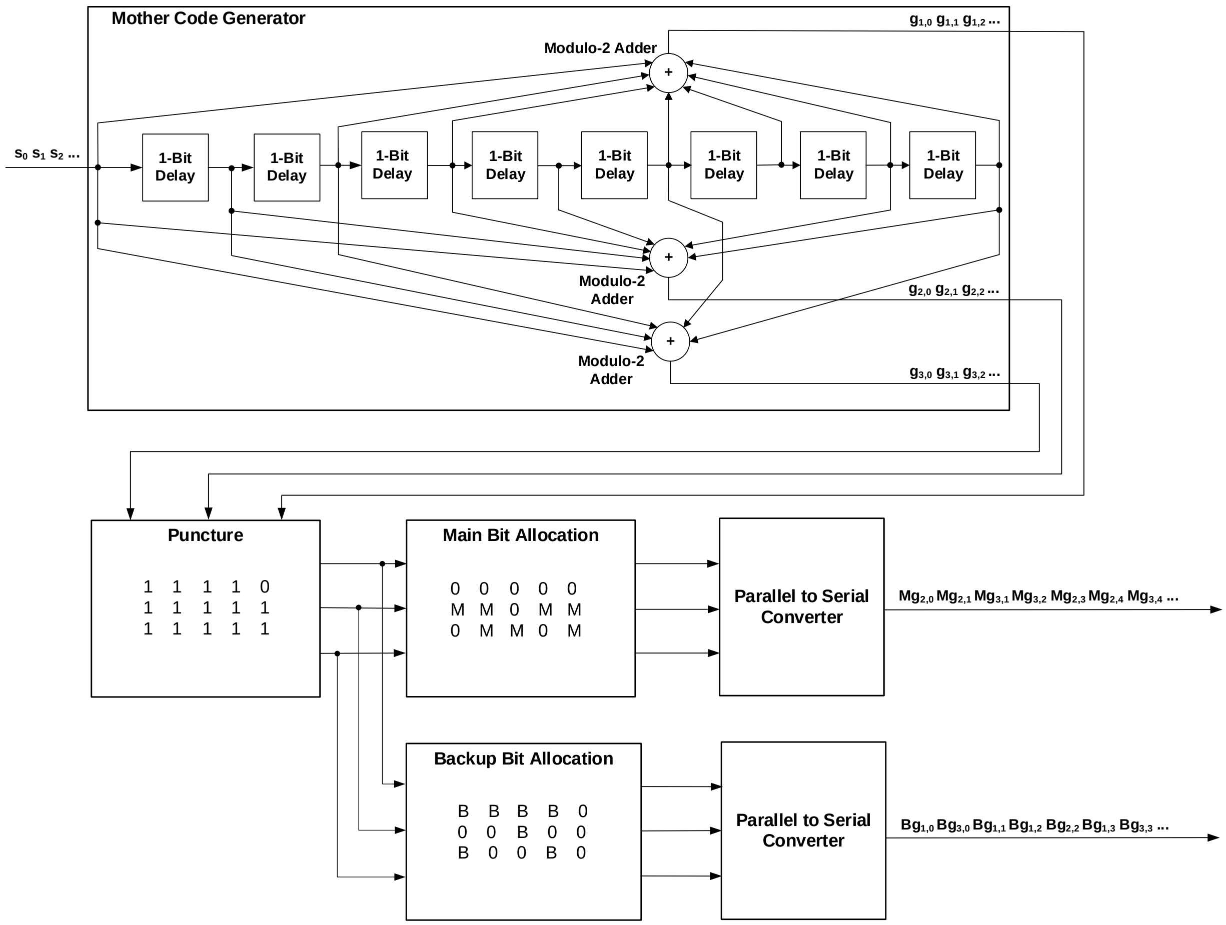


Figure 9-7: Convolutional Encoder – Rate 5/14 Code

9.4 Channel Encoding Data Flow

The channel encoding process for each logical channel in each service mode is specified in Subsection 9.4.1 through Subsection 9.4.11.

9.4.1 Service Mode MP1

Only P1 and PIDS logical channels are active in service mode MP1. The flow of their transfer frames through the channel encoding process for service mode MP1 is shown in Figure 9-8.

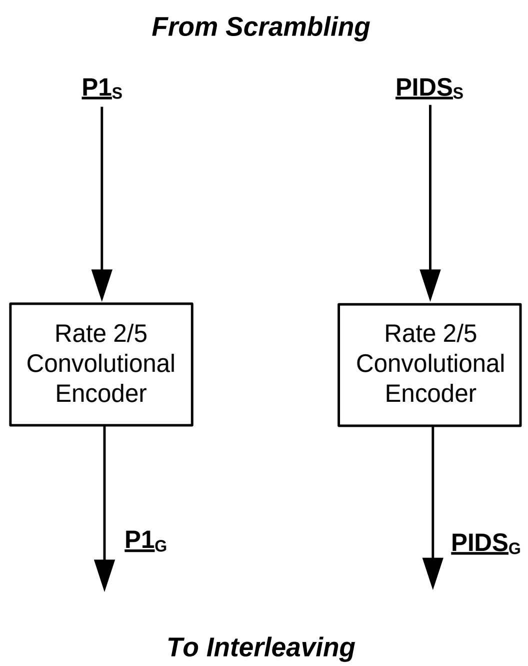


Figure 9-8: Channel Encoding – Service Mode MP1

9.4.2 Service Modes MP2 and MP3

Only P1, P3, and PIDS logical channels are active in service modes MP2 and MP3. The flow of their transfer frames through the channel encoding process for service modes MP2 and MP3 is shown in Figure 9-9.

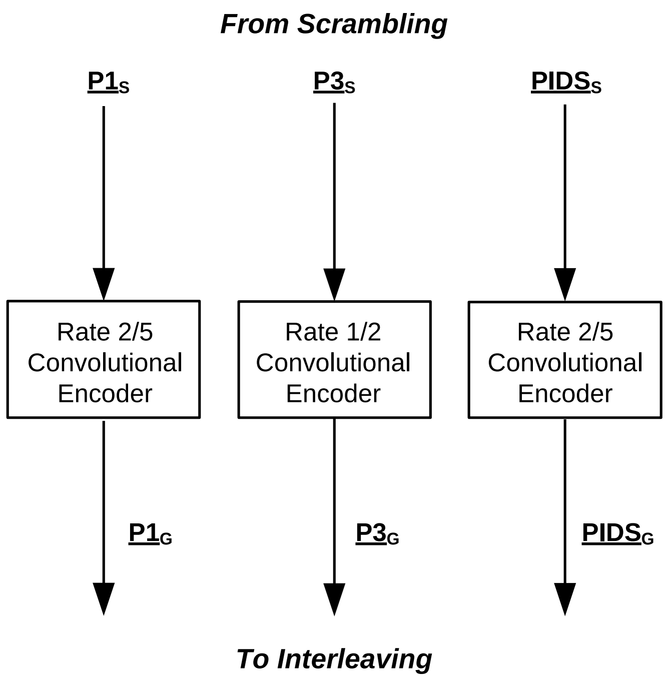


Figure 9-9: Channel Encoding – Service Modes MP2 and MP3

9.4.3 Service Mode MP11

Only P1, P3, P4, and PIDS logical channels are active in service mode MP11. The flow of their transfer frames through the channel encoding process for service mode MP11 is shown in Figure 9-10.

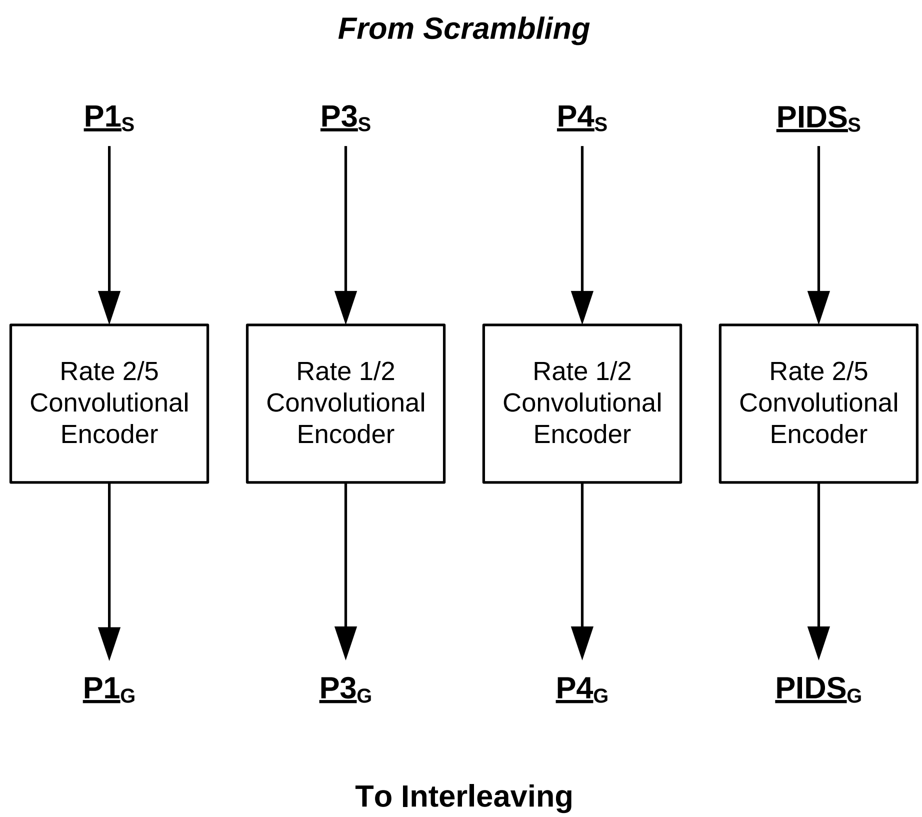


Figure 9-10: Channel Encoding – Service Mode MP11

9.4.4 Service Mode MP5

Only P1, P2, P3, and PIDS logical channels are active in service mode MP5. The flow of their transfer frames through the channel encoding process for service mode MP5 is shown in Figure 9-11.

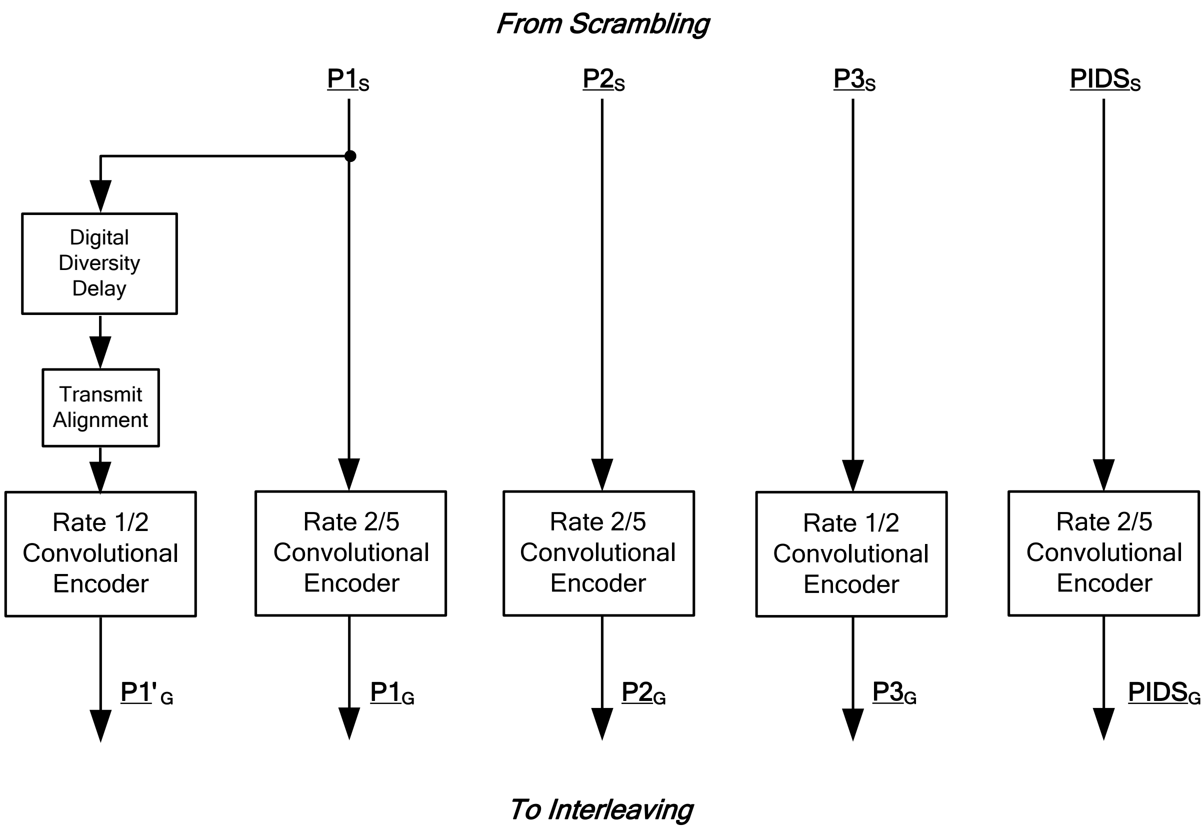


Figure 9-11: Channel Encoding – Service Mode MP5

9.4.5 Service Mode MP6

Only P1, P2, and PIDS logical channels are active in service mode MP6. The flow of their transfer frames through the channel encoding process for service mode MP6 is shown in Figure 9-12.

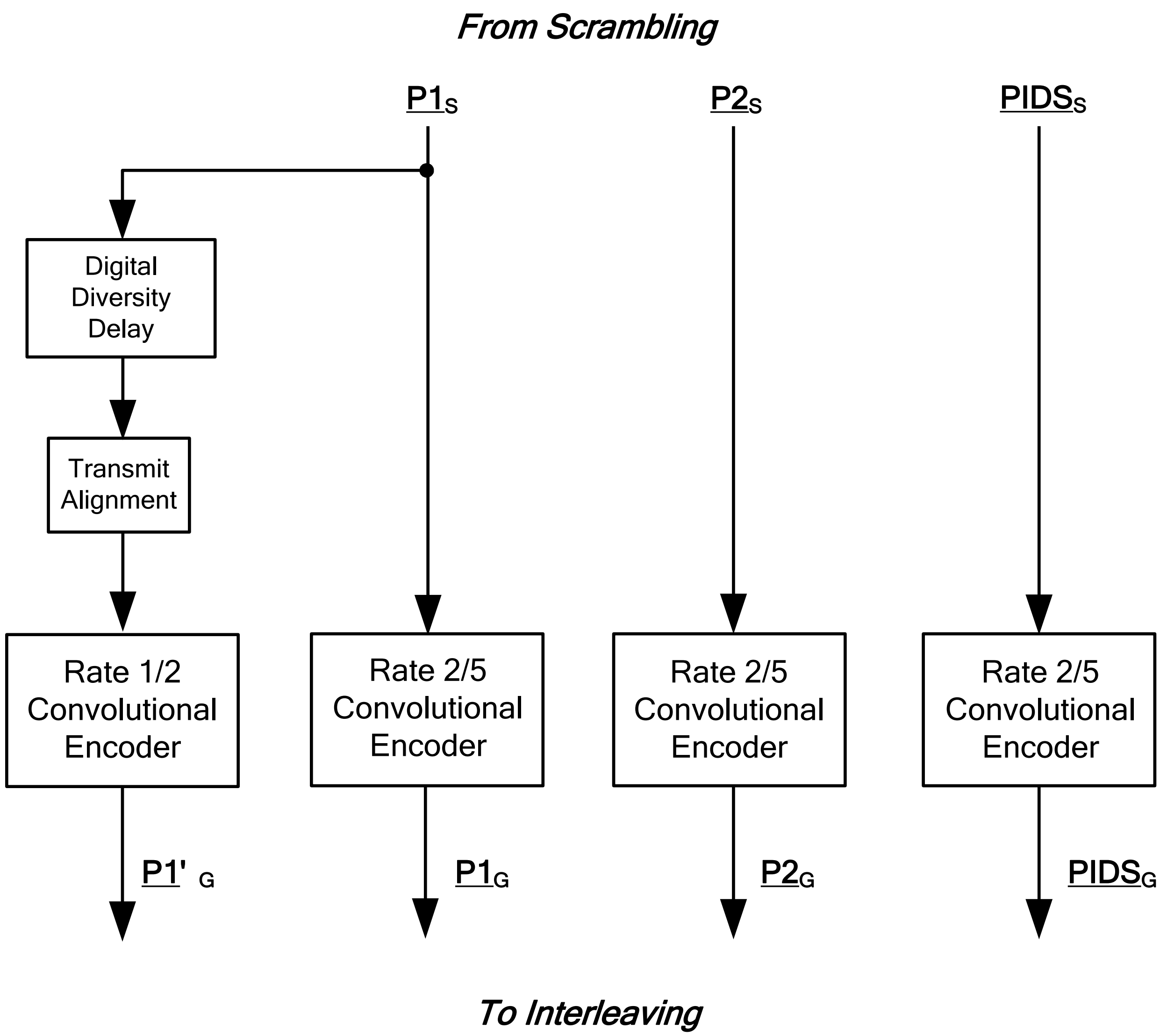


Figure 9-12: Channel Encoding – Service Mode MP6

9.4.6 Service Mode MP1X

Only P1, P4, and PIDS logical channels are active in service mode MP1X. The flow of their transfer frames through the channel encoding process for service mode MP1X is shown in Figure 9-13.

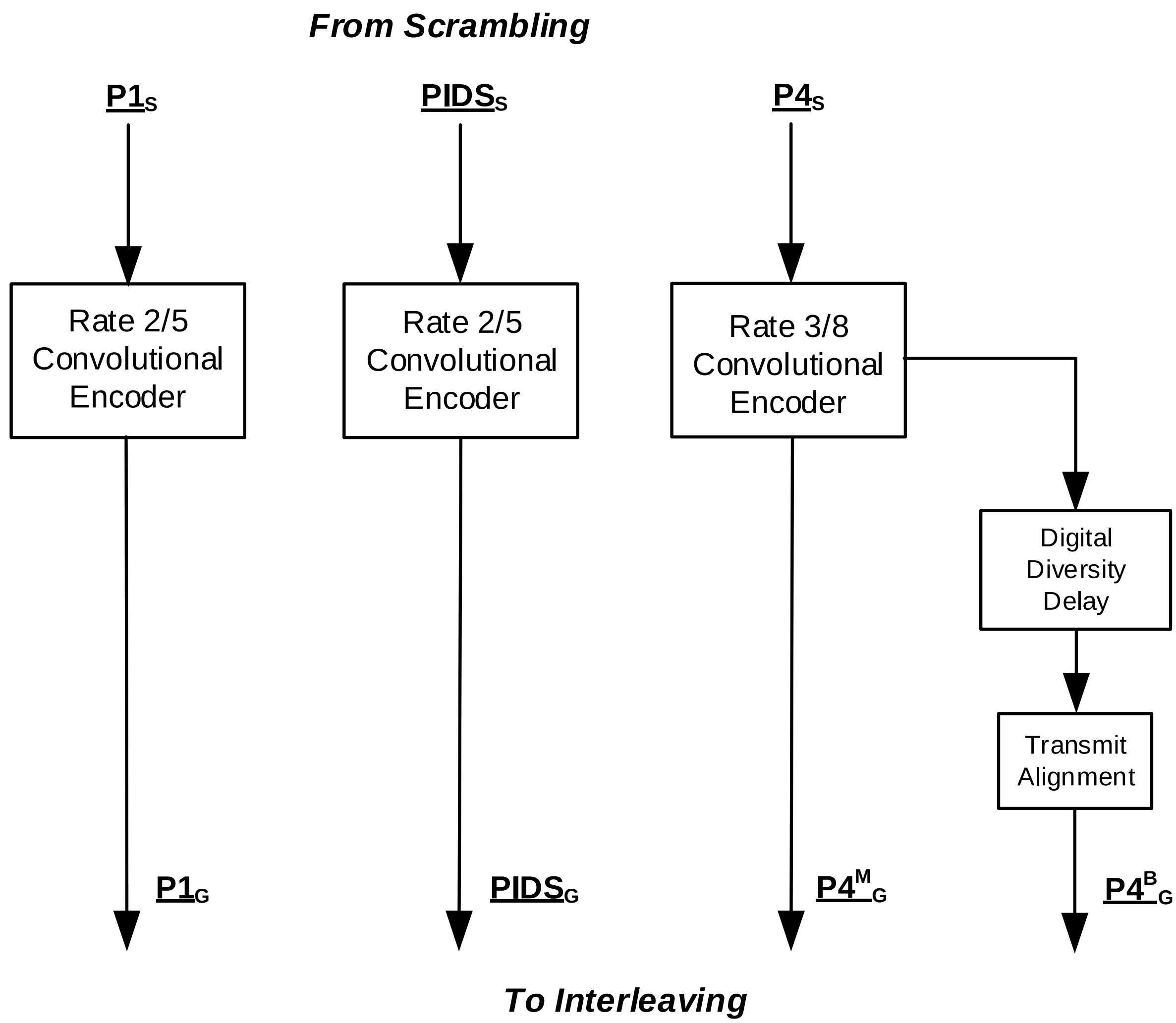


Figure 9-13: Channel Encoding – Service Mode MP1X

9.4.7 Service Mode DSB1

Only P1and PIDS logical channels are active in service mode DSB1. The flow of their transfer frames through the channel encoding process for service mode DSB1 is shown in Figure 9-14.

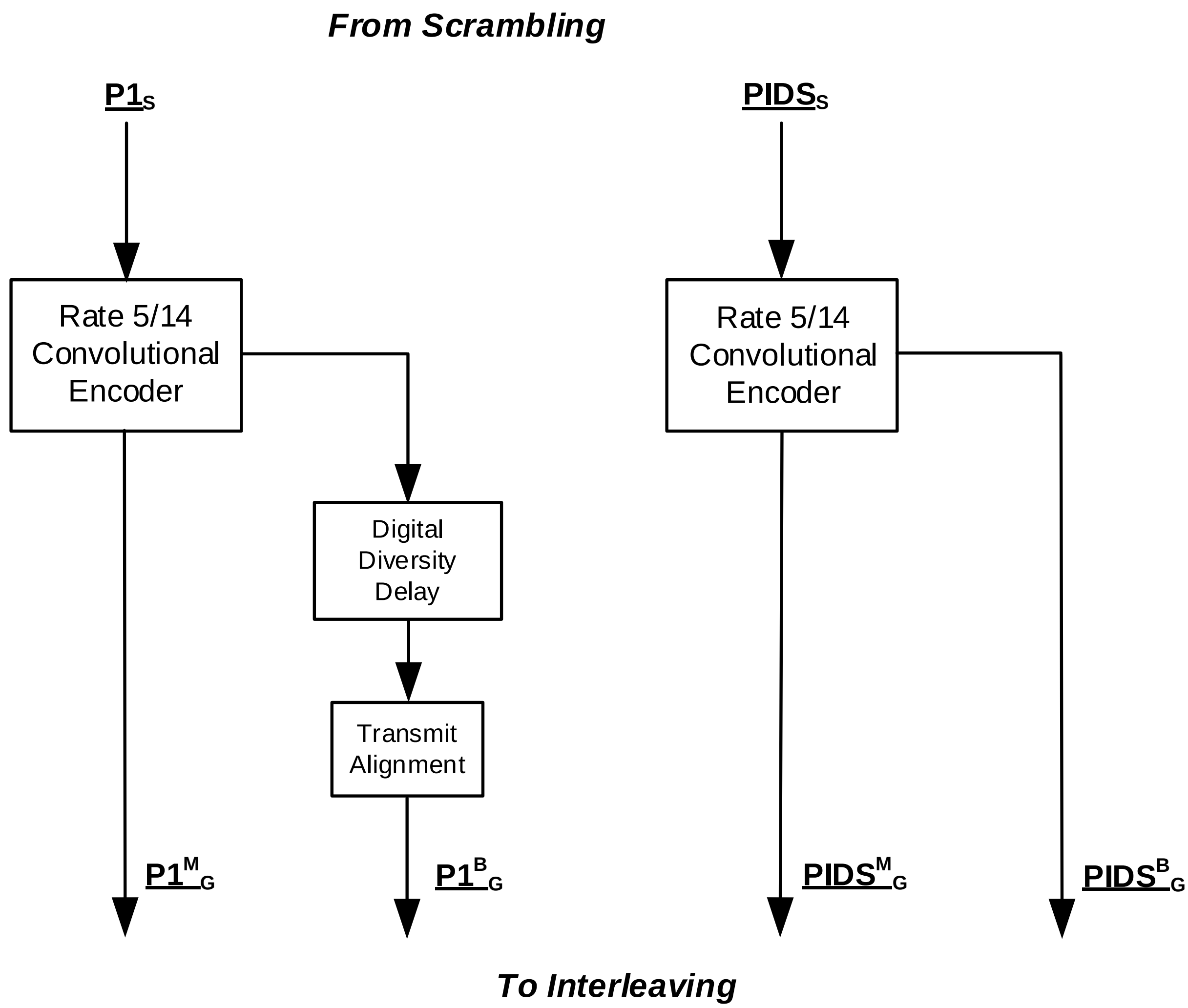


Figure 9-14: Channel Encoding – Service Mode DSB1

9.4.8 Service Mode MP1XOV

Only P1, P4, POV, PIDS, and PIDSOV logical channels are active in service mode MP1XOV. The flow of their transfer frames through the channel encoding process for service mode MP1XOV is shown in Figure 9-15.

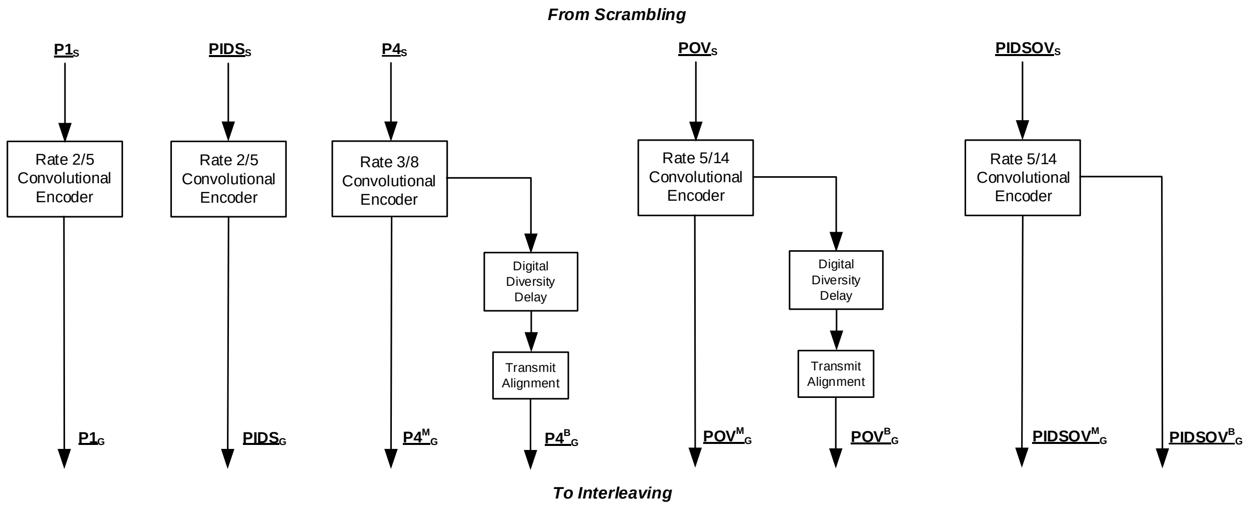


Figure 9-15: Channel Encoding – Service Mode MP1XOV

9.4.9 Service Mode MP6OV

Only P1, P2, POV, PIDS, and PIDSOV logical channels are active in service mode MP6OV. The flow of their transfer frames through the channel encoding process for service mode MP6OV is shown in Figure 9-16.

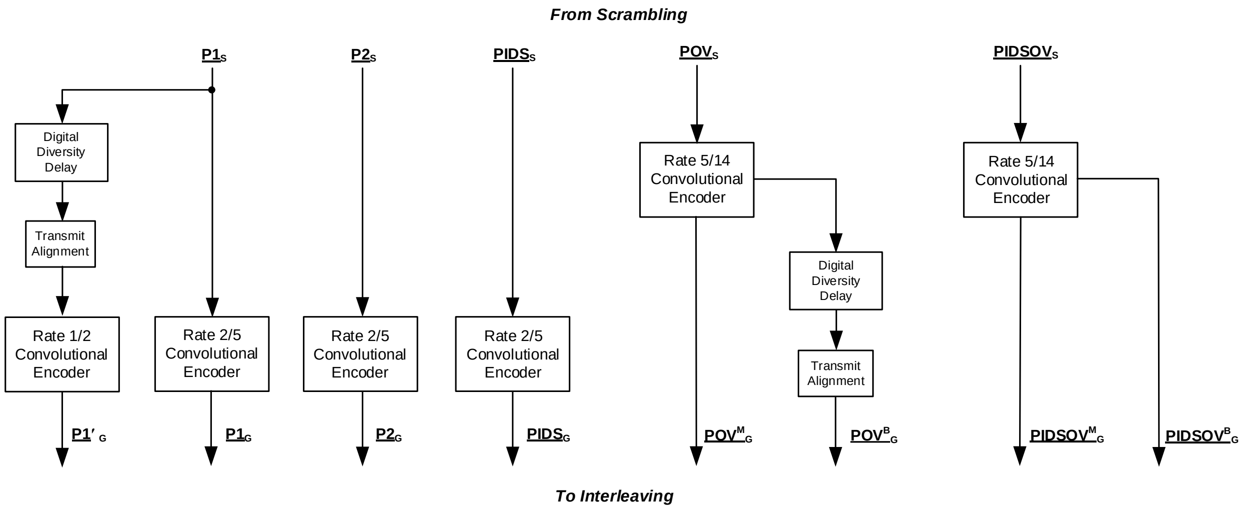


Figure 9-16: Channel Encoding – Service Mode MP6OV

9.4.10 Service Mode DSB1OV

Only P1, POV, PIDS, and PIDSOV logical channels are active in service mode DSB1OV. The flow of their transfer frames through the channel encoding process for service mode DSB1OV is shown in Figure 9-17.

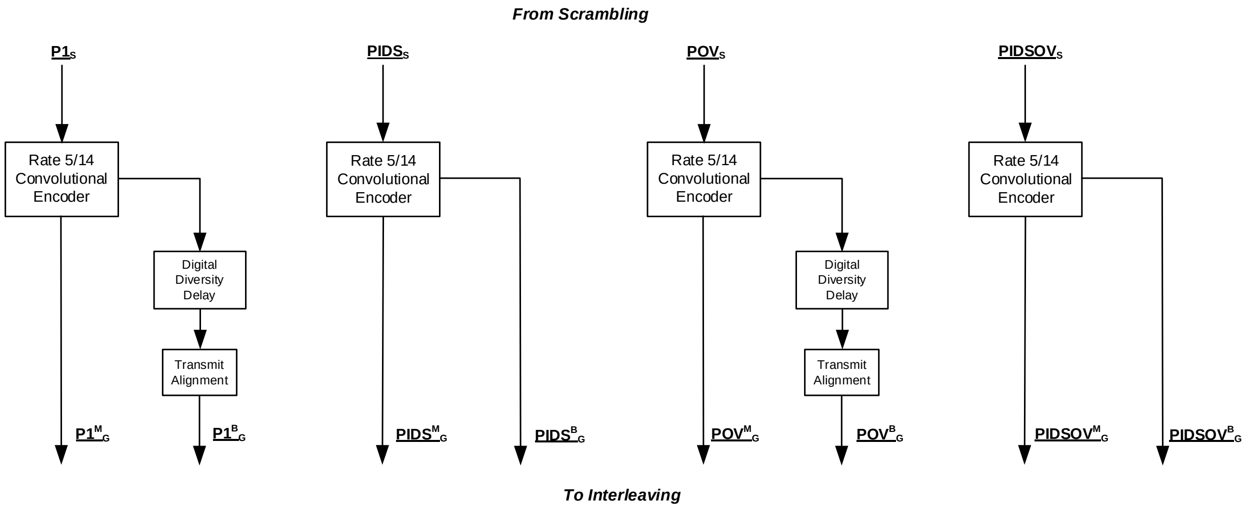


Figure 9-17: Channel Encoding – Service Mode DSB1OV

9.4.11 Service Mode MS5

Only S1 and SIDS logical channels are active in service mode MS5. The flow of their transfer frames through the channel encoding process for service mode MS5 is shown in Figure 9-18.

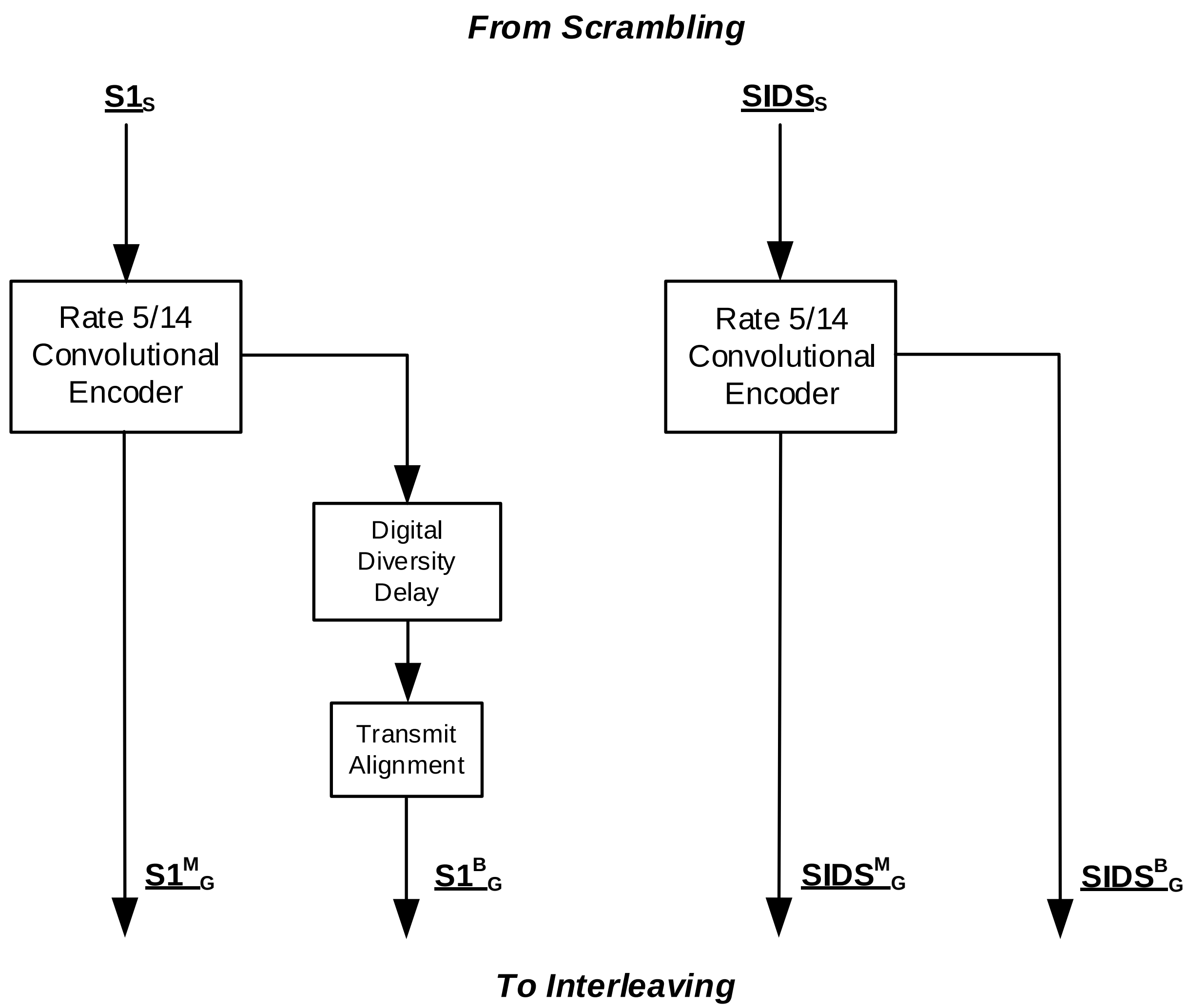


Figure 9-18: Channel Encoding – Service Mode MS5

10 Interleaving

10.1 Introduction

Interleaving consists of four parallel *interleaving processes* (IPs): PM, PX, PB, and SB (see Figure 10-1). An IP contains one or more interleavers, and, in some cases, a *transfer frame multiplexer*.

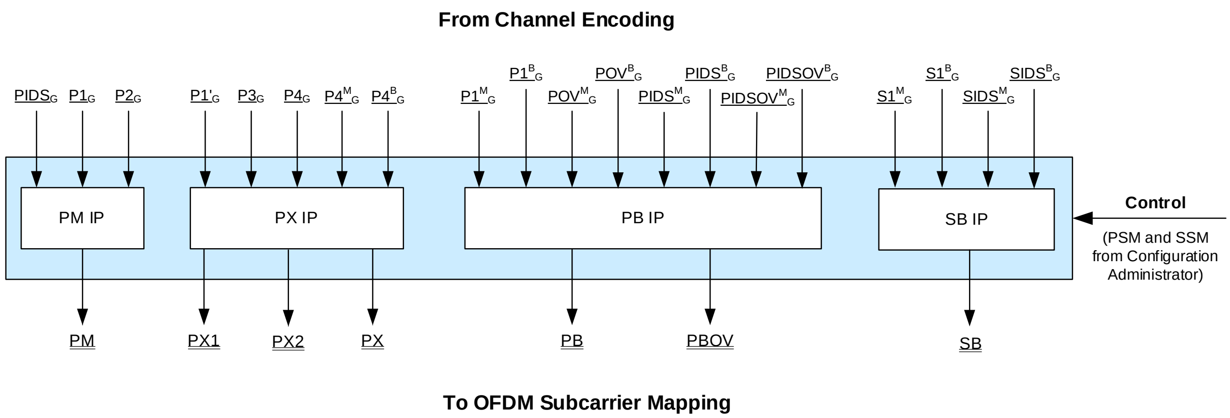


Figure 10-1: Interleaving Conceptual Block Diagram

The service mode determines which inputs and IPs are active at any given time. The universe of inputs for interleaving are the channel-encoded transfer frames from the primary logical channels P1 through P4, POV, PIDS, and PIDSOV and the secondary logical channels S1 and SIDS – including their main and backup components in advanced service modes. Table 10-1 through Table 10-14 show the active IP inputs for each service mode. These tables define the size and rate of the transfer frames on each active logical channel along with the destination interleaver matrix and the number of transfer frames required to fill the destination interleaver matrix.

As shown in Table 10-1 through Table 10-5, although there is one transfer frame per PX1 or PX2 interleaver matrix for the P3 or P4 logical channel, respectively, the interleaver depth is actually two L1 frames. In this case, an internal interleaver matrix is introduced to account for the additional span of the interleaver. Refer to Subsection 10.2.2.2 for details.

The main and backup components of logical channels in advanced service modes are destined for separate main and backup interleavers. After interleaving, these components are then combined into a single interleaver matrix.

Interleaver matrices of bits from all active parallel IPs are transferred to OFDM Subcarrier Mapping, which maps a row of bits from each interleaver partition to frequency partitions. For standard service modes, this is a trivial 1:1 mapping because QPSK modulation is applied to all OFDM data subcarriers. However, for those frequency partitions in advanced service modes in which 16-QAM or 64-QAM is used, there are two or three interleaver partitions, respectively, mapped to each frequency partition. OFDM Subcarrier Mapping is described in detail in Section 12.

Table 10-1: Transfer Frame Characteristics – Service Mode MP1

Logical Channel	Transfer Frame Size (bits)	Transfer Frame Rate (Hz)	Interleaver Matrix	Transfer Frames per Interleaver Matrix
P1	365440	R_f	PM	1
PIDS	200	R_b	PM	16

Table 10-2: Transfer Frame Characteristics – Service Mode MP2

Logical Channel	Transfer Frame Size (bits)	Transfer Frame Rate (Hz)	Interleaver Matrix	Transfer Frames per Interleaver Matrix
P1	365440	R_f	PM	1
PIDS	200	R_b	PM	16
P3	4608	R_p	PX1	1

Table 10-3: Transfer Frame Characteristics – Service Mode MP3

Logical Channel	Transfer Frame Size (bits)	Transfer Frame Rate (Hz)	Interleaver Matrix	Transfer Frames per Interleaver Matrix
P1	365440	R_f	PM	1
PIDS	200	R_b	PM	16
P3	9216	R_p	PX1	1

Table 10-4: Transfer Frame Characteristics – Service Mode MP11

Logical Channel	Transfer Frame Size (bits)	Transfer Frame Rate (Hz)	Interleaver Matrix	Transfer Frames per Interleaver Matrix
P1	365440	R_f	PM	1
PIDS	200	R_b	PM	16
P3	9216	R_p	PX1	1
P4	9216	R_p	PX2	1

Table 10-5: Transfer Frame Characteristics – Service Mode MP5

Logical Channel	Transfer Frame Size (bits)	Transfer Frame Rate (Hz)	Interleaver Matrix	Transfer Frames per Interleaver Matrix
P1	11520	R_p	PM	8
P2	273280	R_f	PM	1
PIDS	200	R_b	PM	16
P1′	9216	R_p	PX2	1
P3	9216	R_p	PX1	1

Table 10-6: Transfer Frame Characteristics – Service Mode MP6

Logical Channel	Transfer Frame Size (bits)	Transfer Frame Rate (Hz)	Interleaver Matrix	Transfer Frames per Interleaver Matrix
P1	23040	R_p	PM	8
P2	181120	R_f	PM	1
PIDS	200	R_b	PM	16
P1′	18432	R_p	PX2	1

Table 10-7: Transfer Frame Characteristics – Service Mode MP1X

Logical Channel	Transfer Frame Size (bits)	Transfer Frame Rate (Hz)	Interleaver Matrix	Transfer Frames per Interleaver Matrix
P1	365440	R_f	PM	1
PIDS	200	R_b	PM	16
$P4^M$	18432	R_p	PXM	8
$P4^B$	18432	R_p	PXB	1

Table 10-8: Transfer Frame Characteristics – Service Mode DSB1

Logical Channel	Transfer Frame Size (bits)	Transfer Frame Rate (Hz)	Interleaver Matrix	Transfer Frames per Interleaver Matrix
$P1^M$	63616	R_p	PBM	8
$P1^B$	63616	R_p	PBB	1
$PIDS^M$	448	R_b	PBM	16
$PIDS^B$	448	R_b	PBB	2

Table 10-9: Reserved

Table 10-10: Transfer Frame Characteristics – Service Mode MP1XOV

Logical Channel	Transfer Frame Size (bits)	Transfer Frame Rate (Hz)	Interleaver Matrix	Transfer Frames per Interleaver Matrix
P1	365440	R_f	PM	1
PIDS	200	R_b	PM	16
$P4^M$	9216	R_p	PXM	8
$P4^B$	9216	R_p	PXB	1
POV^M	31808	R_p	PBOVM	8
POV^B	31808	R_p	PBOVB	1
$PIDSOV^M$	224	R_b	PBOVM	16
$PIDSOV^B$	224	R_b	PBOVB	2

Table 10-11: Transfer Frame Characteristics – Service Mode MP6OV

Logical Channel	Transfer Frame Size (bits)	Transfer Frame Rate (Hz)	Interleaver Matrix	Transfer Frames per Interleaver Matrix
P1	23040	R_p	PM	8
P2	181120	R_f	PM	1
PIDS	200	R_b	PM	16
P1′	18432	R_p	PX2	1
POV^M	31808	R_p	PBOVM	8
POV^B	31808	R_p	PBOVB	1
$PIDSOV^M$	224	R_b	PBOVM	16
$PIDSOV^B$	224	R_b	PBOVB	2

Table 10-12: Transfer Frame Characteristics – Service Mode DSB1OV

Logical Channel	Transfer Frame Size (bits)	Transfer Frame Rate (Hz)	Interleaver Matrix	Transfer Frames per Interleaver Matrix
$P1^M$	63616	R_p	PBM	8
$P1^B$	63616	R_p	PBB	1
$PIDS^M$	448	R_b	PBM	16
$PIDS^B$	448	R_b	PBB	2
POV^M	31808	R_p	PBOVM	8
POV^B	31808	R_p	PBOVB	1
$PIDSOV^M$	224	R_b	PBOVM	16
$PIDSOV^B$	224	R_b	PBOVB	2

Table 10-13: Reserved

Table 10-14: Transfer Frame Characteristics – Service Mode MS5

Logical Channel	Transfer Frame Size (bits)	Transfer Frame Rate (Hz)	Interleaver Matrix	Transfer Frames per Interleaver Matrix
$S1^M$	31808	R_p	SBM	8
$S1^B$	31808	R_p	SBB	1
$SIDS^M$	224	R_b	SBM	16
$SIDS^B$	224	R_b	SBB	2

10.2 Interleaver

An interleaver is a function that takes a vector of bits as its input and outputs a matrix of reordered bits. The reordering of bits before transmission mitigates the impact of burst errors caused by signal fades and interference.

10.2.1 Interleaver Matrix

The interleaver function uses a two-dimensional matrix to reorder a vector of channel-encoded bits. The interleaver allows individual encoded bits or groups of encoded bits to be directed to a specific *interleaver partition* within the interleaver matrix. An interleaver partition can be viewed as a smaller independent interleaver.

Figure 10-2 shows the interleaver matrix used by the PM IP. This interleaver matrix contains 20 interleaver partitions. Interleaver partition 0 is highlighted.

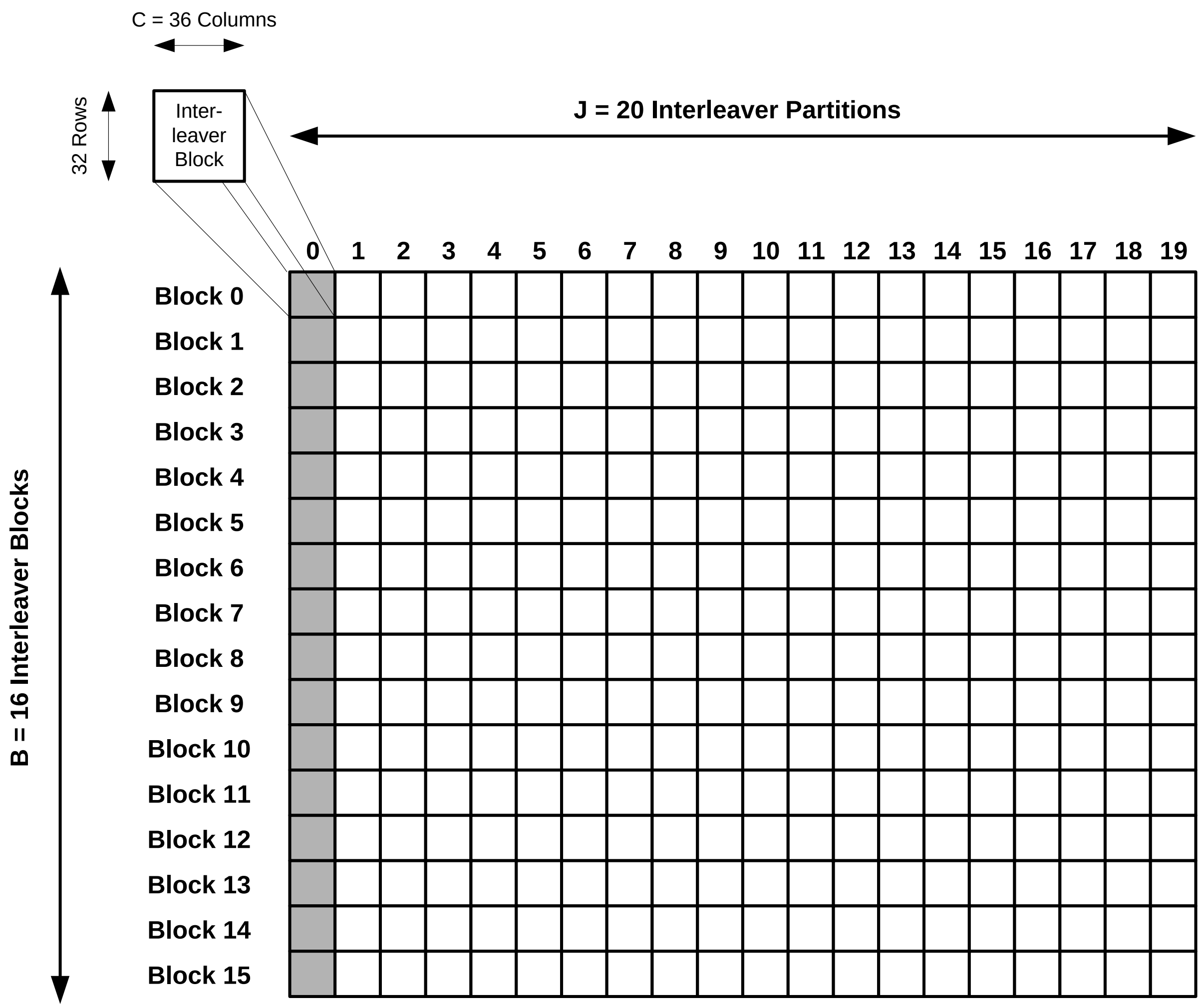


Figure 10-2: PM Interleaver Matrix

In general, the interleaver matrix is divided into J interleaver partitions. Each interleaver partition is divided into B *interleaver blocks*. An interleaver block spans 32 rows and C columns; thus, the dimensions for each interleaver partition in a given interleaver matrix are defined by the expression:

$$(B \cdot 32) \times C$$

For a given interleaver within an IP, the interleaver matrix size can vary with service mode.

10.2.2 Interleaver Computations

The input to each interleaver is a vector of channel encoded bits indexed from i = 0,1,2,…,N-1.

The output of each interleaver is a matrix of bits destined for OFDM Subcarrier Mapping; this matrix has dimensions that are defined by the following expression:

$$(B \cdot 32) \times (J \cdot C)$$

The mapping of each encoded bit to a location in the interleaver matrix is calculated using a set of equations. There are three sets of equations and thus three interleaver types: Interleaver I, Interleaver II, and Interleaver IV (note that Interleaver III is no longer supported). All three interleavers use the variable parameters shown in Table 10-15, except as noted.

Table 10-15: Interleaver Parameters

Interleaver Parameter	Interleaver Parameter Definition
J	The number of interleaver partitions per interleaver matrix.
B	The number of interleaver blocks per interleaver partition.
C	The number of columns per interleaver block.
M	Factor used in interleaver partition assignment calculation (Interleavers I and IV).
$\underline{v}$	Partition assignment vector used to control the relative ordering of interleaver partitions in the interleaver matrix.
b	Number of bits per transfer frame (Interleavers II and IV).
I_0	Index offset value used in k_i calculation (Interleaver II).
N	The number of bits per interleaver input sequence. May span multiple transfer frames.

10.2.2.1 Equation Sets I or II

For a given interleaver using equation set I or II, the steps needed to direct each encoded bit of an input sequence of length N to an interleaver matrix location are as follows:

1. Determine which set of interleaver equations to use by inspecting the IP figures in Subsection 10.4.
2. Assign values to parameters J, B, C, M, $\underline{v}$, b, I_0, and N using the tables in Subsection 10.4.
3. For each i = 0 to N-1, calculate $partition_i$, $block_i$, k_i, row(k_i), and column(k_i). Write the i^{th} input bit to this location in the interleaver matrix.

10.2.2.2 Equation Set IV

Equation set IV implements a convolutional interleaver. With a convolutional interleaver, each write to the interleaver matrix must be followed by a read from the interleaver matrix. Since the total number of bits being interleaved is greater than the transfer frame size, an additional matrix is needed to manage this flow. Thus, the terminology associated with Interleaver IV is as follows:

- Internal interleaver matrix

  The internal interleaver matrix has the dimensions that are defined by the following expression:

  $$(B \cdot 32) \times (J \cdot C)$$

  Bits are written to the interelaver matrix using interleaver equation set IV and bits are read sequentially across rows. It may take multiple transfer frames to fill this matrix. It is full after N bits have been processed.

- Output interleaver matrix

  The ouput interleaver matrix has the dimensions that are defined by the following expression:

  $$\left(\frac{B}{N/b} \cdot 32 \right) \times (J \cdot C)$$

  The output interelaver matrix contains b interleaved bits read from the internal interleaver matrix. The number of bits in this matrix is equal to the size of the input transfer frame or parameter b. Bits are written to this matrix sequentially across rows starting at row 0, column 0. Note that the number of transfer frames per interleaver matrix equals *N/b*.

For a given interleaver using equation set IV, the steps needed to process each encoded bit of an input sequence of length N are as follows:

1. Assign values to parameters J, B, C, M, $\underline{v}$, b, and N using the tables in Subsection 10.4.
2. Initialize the partition assignment counter vector, $\underline{pt}$, to all zeros. The length of this vector equals J.
3. For each i = 0 to N-1:
   - Write a bit to the internal interleaver matrix using a calculated bit address based on the equations in Subsection 10.2.5.
   - Calculate $partition_i$, fetch $\underline{pt}\,[partition_i]$, and calculate $block_i$, row_i, and $column_i$.
   - Write the i^{th} input bit to this location in the internal interleaver matrix.
   - Read a bit from the following row and column of the internal interleaver matrix:

$$readRow = \mathrm{INT}\left(i/CJ\right)$$

$$readColumn = i \ \mathrm{MOD}\ CJ$$

   - Write the bit read from the internal interleaver matrix to the following row and column of the output interleaver matrix:

$$writeRow = \mathrm{INT}\left(\frac{(i \ \mathrm{MOD}\ b)}{CJ}\right)$$

$$writeColumn = (i \ \mathrm{MOD}\ b)\ \mathrm{MOD}\ CJ$$

   - Increment $\underline{pt}\,[partition_i]$

10.2.3 Interleaver I Equations

Interleaver I is used by all IPs.

10.2.3.1 Interleaver Partition Assignment

Compute an index into v to retrieve the interleaver partition assignment:

$$partIndex_i = INT\left(\frac{i+\left(2\cdot INT\left(\frac{M}{4}\right)\right)}{M}\right) MOD\ (\mathrm{length}(\underline{v}))$$

$$partition_i = \underline{v}\ [partIndex_i]$$

For standard service modes, either the number of interleaver partitions J or length(v) could be used as the modulus in the above partition index equation. However, since the two quantities can differ in some advanced service modes, only length(v) generally serves as the proper modulus for all standard and advanced service modes.

10.2.3.2 Interleaver Block Assignment within Interleaver Partition

For M = 1:

$$block_i = \left(\mathrm{INT}\left(\frac{i}{J}\right) + \left(partition_i \cdot 7\right)\right) \mathrm{MOD}\ B$$

For M = 2 or 4:

$$block_i = \left(i + \mathrm{INT}\left(\frac{i}{J\cdot B}\right)\right) \mathrm{MOD}\ B$$

10.2.3.3 Row and Column Assignments within Interleaver Block

$$row(k_i) = (k_i \cdot 11)\ \mathrm{MOD}\ 32$$

$$column(k_i) = \left((k_i \cdot 11) + \mathrm{INT}\left(\frac{k_i}{32\cdot 9}\right)\right) \mathrm{MOD}\ C$$

Where the index k_i is defined as: $k_i = \mathrm{INT}\left(\frac{i}{J\cdot B}\right)$

10.2.4 Interleaver II Equations

Interleaver II is used by the PM, PB, and SB IPs. This interleaver is designed to disperse each $\underline{\text{PIDS}}_G$, $\underline{\text{PIDSOV}}_G$, or $\underline{\text{SIDS}}_G$ transfer frame (including any main and backup components) over one interleaver block (and J interleaver partitions) of the same interleaver matrix written to by Interleaver I. In essence, Interleaver II fills in the unpopulated elements ("holes") left behind by Interleaver I. The position of the holes is the same in each interleaver block of the applicable interleaver matrix.

When using Interleaver II, the parameter b is set to the size of one $\underline{\text{PIDS}}_G$, $\underline{\text{PIDSOV}}_G$, or $\underline{\text{SIDS}}_G$ transfer frame. The variable i, however, must range over the total number of $\underline{\text{PIDS}}_G$, $\underline{\text{PIDSOV}}_G$, or $\underline{\text{SIDS}}_G$ bits required to fill all holes of the interleaver matrix.

10.2.4.1 Interleaver Partition Assignment

Compute an index into $\underline{v}$ to retrieve the interleaver partition assignment:

$$partIndex_i = i \text{ MOD length}(\underline{v})$$

$$partition_i = \underline{v}\,[partIndex_i]$$

10.2.4.2 Interleaver Block Assignment within Interleaver Partition

$$block_i = \text{INT}\left(\frac{i}{b}\right)$$

10.2.4.3 Row and Column Assignments within Interleaver Block

$$row(k_i) = (k_i \cdot 11) \text{ MOD } 32$$

$$column(k_i) = \left((k_i \cdot 11) + \text{INT}\left(\frac{k_i}{32 \cdot 9}\right)\right) \text{MOD } C$$

Where the index k_i is defined as: $k_i = \left(\text{INT}\left(\frac{i}{J}\right) \text{MOD}\left(\frac{b}{J}\right)\right) + \left(\frac{I_0}{J \cdot B}\right)$

10.2.5 Interleaver IV Equations

Interleaver IV is used by the PX IP to interleave $\underline{P3}_G$ and $\underline{P4}_G$ transfer frames. The pertinent equations are presented in Subsection 10.2.5.1 through Subsection 10.2.5.4.

Define a supporting parameter which represents the number of bits in an interleaver block:

$$Bk\_bits = 32 \cdot C$$

Define a second supporting parameter:

$$Bk\_adj = 32 \cdot C - 1$$

10.2.5.1 Interleaver Partition Assignment

Compute an index into $\underline{v}$ to retrieve the interleaver partition assignment:

$$partIndex_i = \mathrm{INT}\left(\frac{i + \left(2 \cdot \mathrm{INT}\left(\frac{M}{4}\right)\right)}{M}\right) \mathrm{MOD}\ \left(length(\underline{v})\right)$$

$$partition_i = \underline{v}\ [partIndex_i]$$

Define a vector of partition assignment counters, $\underline{pt}$, whose length is equal to the number of partitions. Fetch the appropriate counter for $partition_i$:

$$pt_i = \underline{pt}\ [partition_i]$$

The partition assignment counter for a given partition is incremented each time an allocation is made to that partition. The initial value of each of the partition assignment counters is set to 0.

10.2.5.2 Interleaver Block Assignment within Interleaver Partition

Using the applicable parameters, apply the following equation:

$$block_i = \left(pt_i + \left(partition_i \cdot 7\right) - \left(Bk\_adj \cdot \mathrm{INT}\left(\frac{pt_i}{Bk\_bits}\right)\right)\right) \mathrm{MOD}\ B$$

10.2.5.3 Row Assignment within Interleaver Block

Using the applicable parameters, apply the following equation:

$$row_i = \mathrm{INT}\left(\frac{(11 \cdot pt_i)\ \mathrm{MOD}\ Bk\_bits}{C}\right)$$

10.2.5.4 Column Assignment within Interleaver Block

Using the applicable parameters, apply the following equation:

$$column_i = (pt_i \cdot 11)\ \mathrm{MOD}\ C$$

10.3 Transfer Frame Multiplexer

For some IPs, a transfer frame multiplexer is required. For each logical channel, the transfer frame multiplexer collects an integer number of transfer frames. The transfer frame multiplexer then concatenates all accumulated transfer frames into a single vector U. All IPs require transfer frame multiplexers since they intersperse multiple logical channels in a common interleaver over the same row and column spans, or they buffer channel encoder outputs at the block-pair rate for the frame-length main interleaver in advanced service modes.

The transfer frame concatenation ordering at the output of each transfer frame multiplexer is shown in Figure 10-, Figure 10-11, and Figure 10-14. The first bit of the first transfer frame becomes the first bit of U. The first bit of each subsequent transfer frame follows the last bit of the previous transfer frame.

10.4 Interleaving Process Descriptions

This subsection discusses the detailed provisions governing implementation of each IP for every applicable service mode.

10.4.1 PM Interleaving Process

The PM IP interleaves the bits mapped to the Primary Main sidebands depicted in Figure 7-2 through Figure 7-8, Figure 7-11, and Figure 7-12. This IP is active in primary service modes MP1, MP2, MP3, MP11, MP5, MP6, MP1X, MP1XOV, and MP6OV. The PM IP disperses multiple logical channels into a single interleaver matrix, PM.

10.4.1.1 Service Modes MP1 through MP3, MP11, MP1X, and MP1XOV

Figure 10-3 shows the PM IP for service modes MP1 through MP3, MP11, MP1X, and MP1XOV. This IP utilizes two interleavers. These interleavers share a common interleaver output matrix, PM. The inputs to the PM IP are the $P1_G$ and $PIDS_G$ transfer frames. The number of transfer frames required to fill the interleaver matrix are shown in Table 10-1 through Table 10-4, Table 10-7, and Table 10-10 for service modes MP1 through MP3, MP11, MP1X, and MP1XOV, respectively.

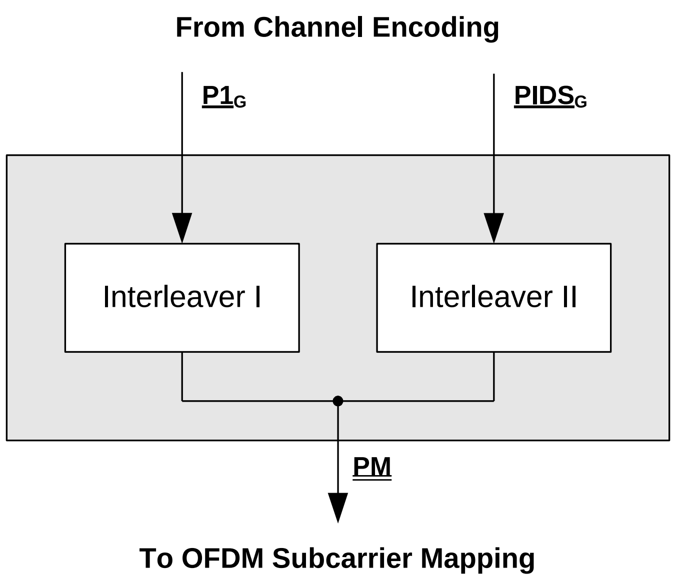


Figure 10-3: PM IP – Service Modes MP1 through MP3, MP11, MP1X, and MP1XOV

The interleaving process must maintain a specific transfer frame alignment and synchronization at its output.

For a given logical channel, the BC range *m1:m2* indicates which L1 blocks are spanned by the designated transfer frame. The ALFN *n* is the absolute L1 frame number.

The steps required to process the IP inputs for L1 frame n are given as follows:

1. ***Interleave $\underline{P1}_G$ transfer frame***
   The vector $\underline{P1}_G$ is interleaved into $\underline{PM}$ using Interleaver I with the parameters shown in Table 10-16. The sequence P1 is dispersed over the full row and column span of $\underline{PM}$, leaving holes to be filled by Interleaver II with $\underline{PIDS}_G$ data.

   Table 10-16: PM Interleaver I Parameter Values

J	B	C	M	$\underline{v}$	b	I_0	N
20	16	36	1	See note below	N/A	N/A	365440

   Note: *$\underline{v}$ = [10,2,18,6,14,8,16,0,12,4,11,3,19,7,15,9,17,1,13,5]*

2. ***Interleave $\underline{PIDS}_G$ transfer frames***
   Each $\underline{PIDS}_G$ transfer frame is interleaved into $\underline{PM}$, using Interleaver II with the parameters shown in Table 10-17.

   Table 10-17: PM Interleaver II Parameter Values

J	B	C	M	$\underline{v}$	b	I_0	N
20	16	36	1	See note below	200	365440	3200

   Note: *$\underline{v}$ = [10,2,18,6,14,8,16,0,12,4,11,3,19,7,15,9,17,1,13,5]*

Interleaver II constrains the row span of each interleaved $\underline{PIDS}_G$ transfer frame to one interleaver block (32 rows). This is accomplished by properly setting the interleaver variable i before each execution of Interleaver II. Table 10-18 shows the relationship between the variable i and the BC of the $\underline{PIDS}_G$ transfer frames.

Table 10-18: Bit Numbering of $\underline{PIDS}_G$ Transfer Frames

BC	Range of variable i
0	0…199
1	200…399
2	400…599
3	600…799
4	800…999
5	1000…1199
6	1200…1399
7	1400…1599
8	1600…1799
9	1800…1999
10	2000…2199
11	2200…2399
12	2400…2599
13	2600…2799
14	2800…2999
15	3000…3199

When Interleaver I has processed one vector $\underline{P1}_G$ and Interleaver II has processed one $\underline{PIDS}_G$ transfer frame, a 32 x J *submatrix* of $\underline{\underline{PM}}$ is completely full and ready for OFDM Subcarrier Mapping. Each successive $\underline{PIDS}_G$ transfer frame is interleaved over the next successive interleaver block (for J interleaver partitions). After each $\underline{PIDS}_G$ transfer frame is processed by Interleaver II, the next 32 x J submatrix of $\underline{\underline{PM}}$ is available for OFDM Subcarrier Mapping.

After Interleaver I has processed one vector $\underline{P1}_G$ and Interleaver II has processed sixteen $\underline{PIDS}_G$ transfer frames, $\underline{\underline{PM}}$ is completely filled and the processing flow resets.

10.4.1.2 Service Modes MP5, MP6, and MP6OV

Figure 10-4 shows the PM IP for service modes MP5, MP6, and MP6OV. The PM processing in MP5, MP6, and MP6OV is very similar to that of MP1 through MP3, MP11, MP1X, and MP1XOV. The difference lies in the number of $\underline{P1}_G$ transfer frames required to fill the interleaver. The $\underline{P2}_G$ transfer frame is now multiplexed with $\underline{P1}_G$. As shown in Table 10-5, Table 10-6, and Table 10-11, the ratio of $\underline{P1}_G$ to $\underline{P2}_G$ transfer frames is 8:1. Before Interleaver I is invoked, the $\underline{P1}_G$ and $\underline{P2}_G$ transfer frames are multiplexed into the vector $\underline{U}$ in the manner shown in Figure 10-5.

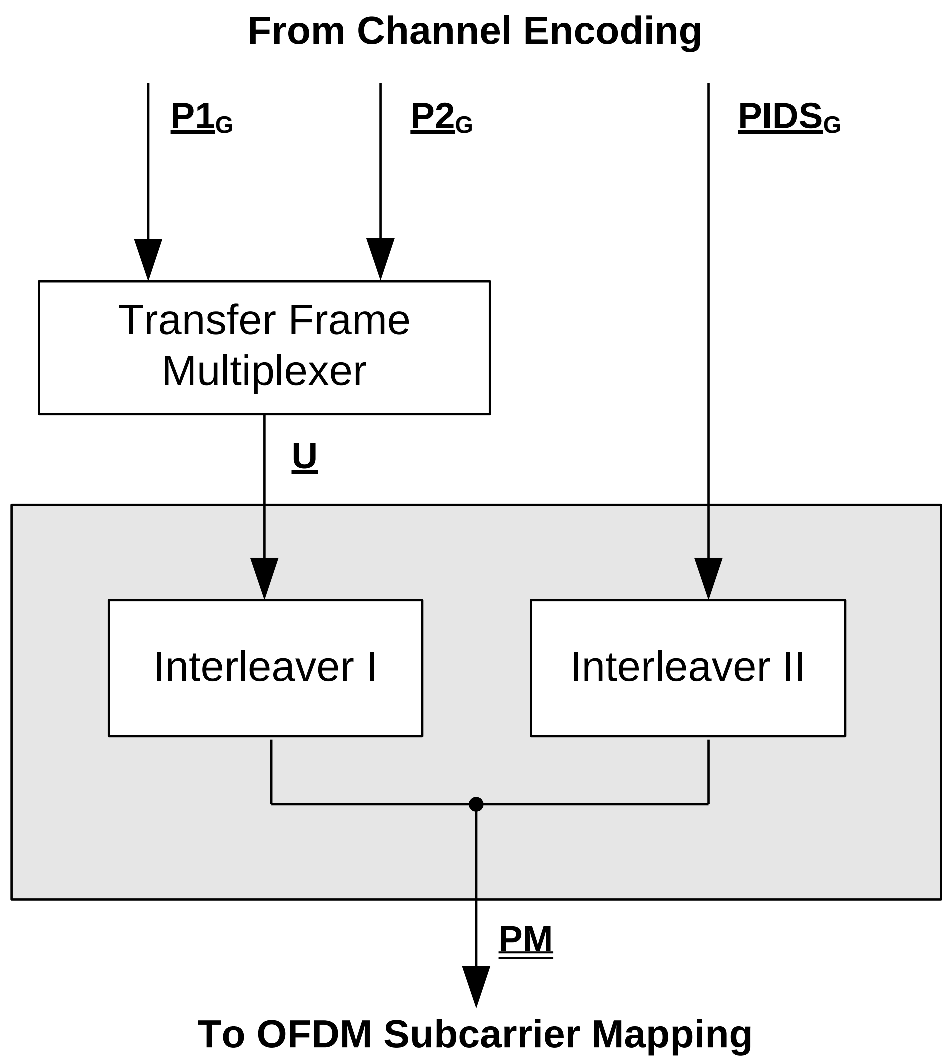


Figure 10-4: PM IP – Service Modes MP5, MP6, and MP6OV

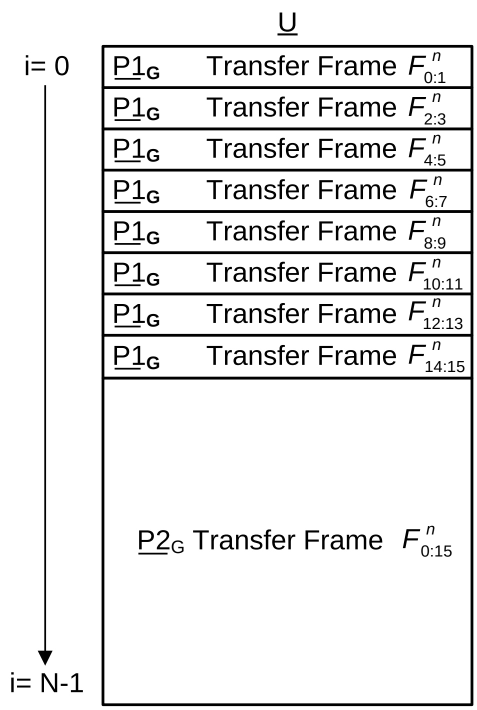


Figure 10-5: PM Transfer Frame Multiplexer Output – Service Modes MP5, MP6, and MP6OV

All processing details subsequent to the transfer frame multiplexer are identical to those described in Subsection 10.4.1.1.

10.4.2 PX Interleaving Process

The PX IP interleaves bits destined for the Primary Extended sidebands shown in Figure 7-3 through Figure 7-8, Figure 7-11, and Figure 7-12. This IP is active in primary service modes MP2, MP3, MP5, MP6, MP11, MP1X, MP1XOV, and MP6OV. Interleaver matrices PX1, PX2, or PX are active. In service modes MP2 and MP3, only PX1 is active. In service modes MP6 and MP6OV, only PX2 is active. In service modes MP5 and MP11, both PX1 and PX2 are active. In service modes MP1X and MP1XOV, only PX is active. $P3_G$ transfer frames are interleaved into PX1, $P1'_G$ or $P4_G$ transfer frames are interleaved into PX2, and $P4^M{}_G$ and $P4^B{}_G$ transfer frames are interleaved into PX.

A long convolutional interleaver is applied using Interleaver IV. A single transfer frame fills the PX1 and PX2 interleaver matrices, as indicated by Table 10-2 through Table 10-6 and Table 10-11. However, multiple $P4^M{}_G$ transfer frames are required to fill the PX interleaver matrix, as indicated by Table 10-7 and Table 10-10.

10.4.2.1 Service Modes MP2 and MP3

Figure 10- shows the PX IP for service modes MP2 and MP3. In these service modes, the PX IP interleaves $P3_G$ transfer frames into an internal interleaver matrix and outputs them to PX1 (the output interleaver matrix) using Interleaver IV. The service mode dependent Interleaver IV parameter values are shown in Table 10-19. Although the transfer frame rate is common, the size of the $P3_G$ transfer frames varies with service mode. Consequently, the number of interleaver partitions in the PX1 interleaver matrix also varies.

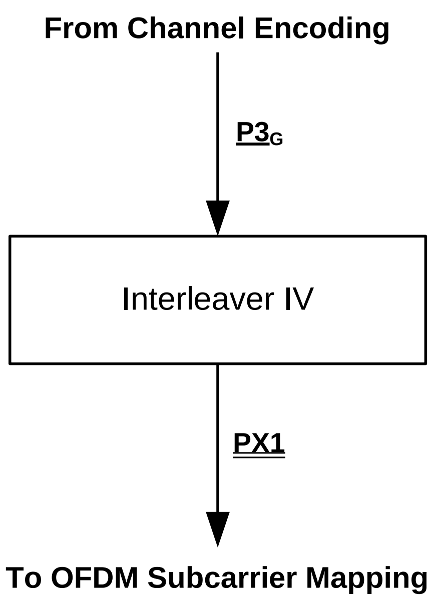


Figure 10-6: PX IP – Service Modes MP2 and MP3

Table 10-19: PX1 Interleaver IV Parameter Values – Service Modes MP2 and MP3

Service Mode	J	B	C	M	V	b	I_0	N
MP2	2	32	36	4	[0,1]	4608	N/A	73728
MP3	4	32	36	2	[0,1,2,3]	9216	N/A	147456

Although the size of the internal interleaver matrix used by Interleaver IV is 16 $P3_G$ transfer frames, Interleaver IV is described as processing one $P3_G$ transfer frame at a time. Every time a bit is written to the internal interleaver matrix used by Interleaver IV, a bit is read sequentially from this matrix and output sequentially to PX1. The size of PX1 is equal to the length of one $P3_G$ transfer frame. Thus, for every $P3_G$ transfer frame processed by Interleaver IV, the PX1 output matrix is completely filled. After Interleaver IV has consumed 16 $P3_G$ transfer frames and 16 PX1 matrices have been filled and output, the internal interleaver matrix is completely filled and the processing flow resets.

In practical applications, because the interleaver is convolutional, the number of bits input to and output from Interleaver IV can be any length less than or equal to N, the capacity of the internal interleaver matrix. The concept of an internal interleaver matrix is described here for notational convenience.

10.4.2.2 Service Mode MP11

In service mode MP11, the PX IP consists of two parallel interleavers of type Interleaver IV. One interleaver processes $\underline{P3}_G$ transfer frames and outputs them to $\underline{PX1}$, and the other processes $\underline{P4}_G$ transfer frames and outputs them to $\underline{PX2}$, as shown in Figure 10-7.

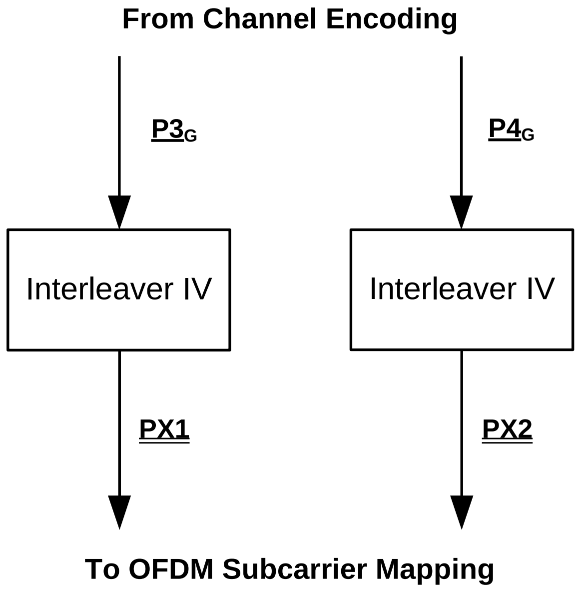


Figure 10-7: PX IP – Service Mode MP11

Because there are two convolutional interleavers, the PX IP uses two internal interleaver matrices in the manner described in Subsection 10.4.2.1. $\underline{P3}_G$ transfer frames are interleaved into the internal interleaver matrix of Interleaver IV Instance 1. $\underline{P4}_G$ transfer frames are interleaved into the internal interleaver matrix of Interleaver IV Instance 2. These processes are synchronized. Both instances of Interleaver IV are configured with the parameters shown in Table 10-20.

The description for each of these parallel processes is as described in Subsection 10.4.2.1.

Table 10-20: Interleaver IV Parameter Values – Service Mode MP11

Service Mode	J	B	C	M	$\underline{V}$	b	I_0	N
MP11	4	32	36	2	[0,1,2,3]	9216	N/A	147456

10.4.2.3 Service Mode MP5

In service mode MP5, the PX IP consists of two parallel interleavers, one of type Interleaver I, and the other of type Interleaver IV. Interleaver I processes $\underline{P1'}_G$ transfer frames, and Interleaver IV processes $\underline{P3}_G$ transfer frames. Figure 10-8 shows the PX IP in service mode MP5.

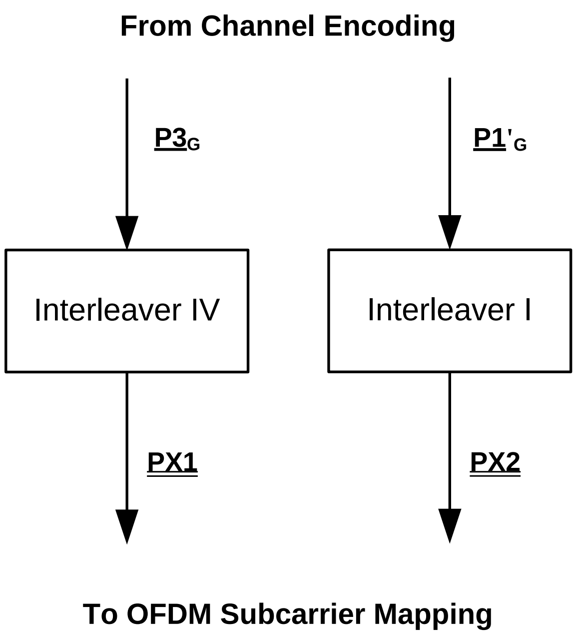


Figure 10-8: PX IP – Service Mode MP5

Interleaver I is configured with the parameters shown in Table 10-21 while Interleaver IV is configured with the parameters shown in Table 10-22.

Table 10-21: PX Interleaver I Parameter Values – Service Mode MP5

Service Mode	J	B	C	M	$\underline{v}$	b	I_0	N
MP5	4	2	36	2	[0,1,2,3]	N/A	N/A	9216

Table 10-22: PX Interleaver IV Parameter Values – Service Mode MP5

Service Mode	J	B	C	M	$\underline{v}$	b	I_0	N
MP5	4	32	36	2	[0,1,2,3]	9216	N/A	147456

After Interleaver I has processed one $\underline{P1'}_G$ transfer frame, PX2 is completely filled and its processing flow resets. After Interleaver IV has consumed 16 $\underline{P3}_G$ transfer frames and output 16 PX1 matrices, its internal interleaver matrix is completely filled and the processing flow resets.

10.4.2.4 Service Modes MP6 and MP6OV

Figure 10-9 shows the PX IP for service modes MP6 and MP6OV. In these service modes, the PX IP interleaves $\underline{P1}'_G$ transfer frames using Interleaver I, configured with the parameter values shown in Table 10-23.

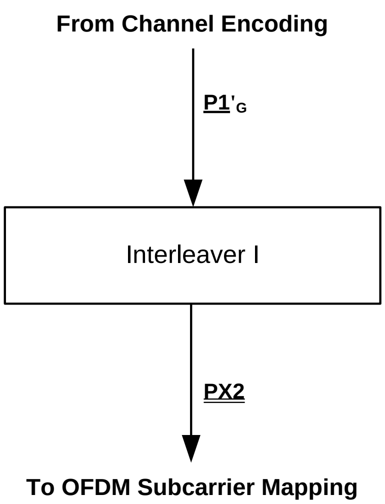


Figure 10-9: PX IP – Service Modes MP6 and MP6OV

Table 10-23: PX2 Interleaver I Parameter Values – Service Modes MP6 and MP6OV

Service Mode	J	B	C	M	$\underline{v}$	b	I_0	N
MP6	8	2	36	1	[0,1,3,2,4,5,7,6]	N/A	N/A	18432

After Interleaver I has processed one $\underline{P1}'_G$ transfer frame, $\underline{\underline{PX2}}$ is completely filled and the processing flow resets.

10.4.2.5 Service Modes MP1X and MP1XOV

Figure 10-10 shows the PX IP for advanced service modes MP1X and MP1XOV. There are three basic processing steps: transfer frame multiplexing, interleaving, and interlacing.

Both $\underline{P4}^{B}_{G}$ and $\underline{P4}^{M}_{G}$ transfer frames arrive from channel encoding at the block-pair rate R_p. Only $\underline{P4}^{M}_{G}$ transfer frames require multiplexing because they are interleaved over an entire L1 frame (16 blocks). The transfer frame multiplexer collects and concatenates eight transfer frames into a single vector $\underline{U}$ prior to interleaving on an L1 frame boundary, as shown in Figure 10-1111. The first bit of the first transfer frame becomes the first bit of $\underline{U}$. The first bit of each subsequent transfer frame follows the last bit of the previous transfer frame.

The PX IP consists of two parallel interleavers of type Interleaver I. $\underline{P4}^{B}_{G}$ transfer frames arrive at the block pair rate and are interleaved directly into the $\underline{\underline{PXB}}$ interleaver matrix immediately upon receipt at each block-pair boundary, completely filling the interleaver with a single transfer frame. Interleaver I parameters for $\underline{\underline{PXB}}$ are shown in Table 10-24 for service modes MP1X and MP1XOV.

The other interleaver processes $\underline{P4}^{M}_{G}$ transfer frames following transfer frame multiplexing. Vector $\underline{U}$ is interleaved into $\underline{\underline{PXM}}$ on an L1 frame boundary. Interleaver I parameters for $\underline{\underline{PXM}}$ are shown in Table 10-25 for service modes MP1X and MP1XOV.

After interleaving, bits from a single block of current active interleaver matrices $\underline{\underline{PXB}}$ and $\underline{\underline{PXM}}$ are *interlaced* to form the $\underline{\underline{PX}}$ interleaver matrix. Interlacing consists of alternately extracting bits, row by row, from the selected block of $\underline{\underline{PXM}}$ and $\underline{\underline{PXB}}$ and writing them into a single $\underline{\underline{PX}}$ interleaver matrix.

The $\underline{\underline{PXB}}$ interleaver matrix has a depth of two blocks, while $\underline{\underline{PXM}}$ has a depth of sixteen blocks. For a given block *B* (ranging from 0 to 15), bits are selected from block *B modulo 2* of the current instance of $\underline{\underline{PXB}}$, and bits are selected from block *B* of the current instance of $\underline{\underline{PXM}}$.

Since there are *C* columns in $\underline{\underline{PXB}}$ and $\underline{\underline{PXM}}$, $\underline{\underline{PX}}$ has dimensions of 32 rows by 2·C columns. Thus, for a given row i (ranging from 0 to 31):

$$\underline{\underline{PX}}(i) = \{\underline{\underline{PXM}}(i,0), \underline{\underline{PXB}}(i,0), \underline{\underline{PXM}}(i,1), \underline{\underline{PXB}}(i,1), \ldots, \underline{\underline{PXM}}(i,C-1), \underline{\underline{PXB}}(i,C-1)\}$$

Main/backup interlacing is illustrated in Figure 10-12.

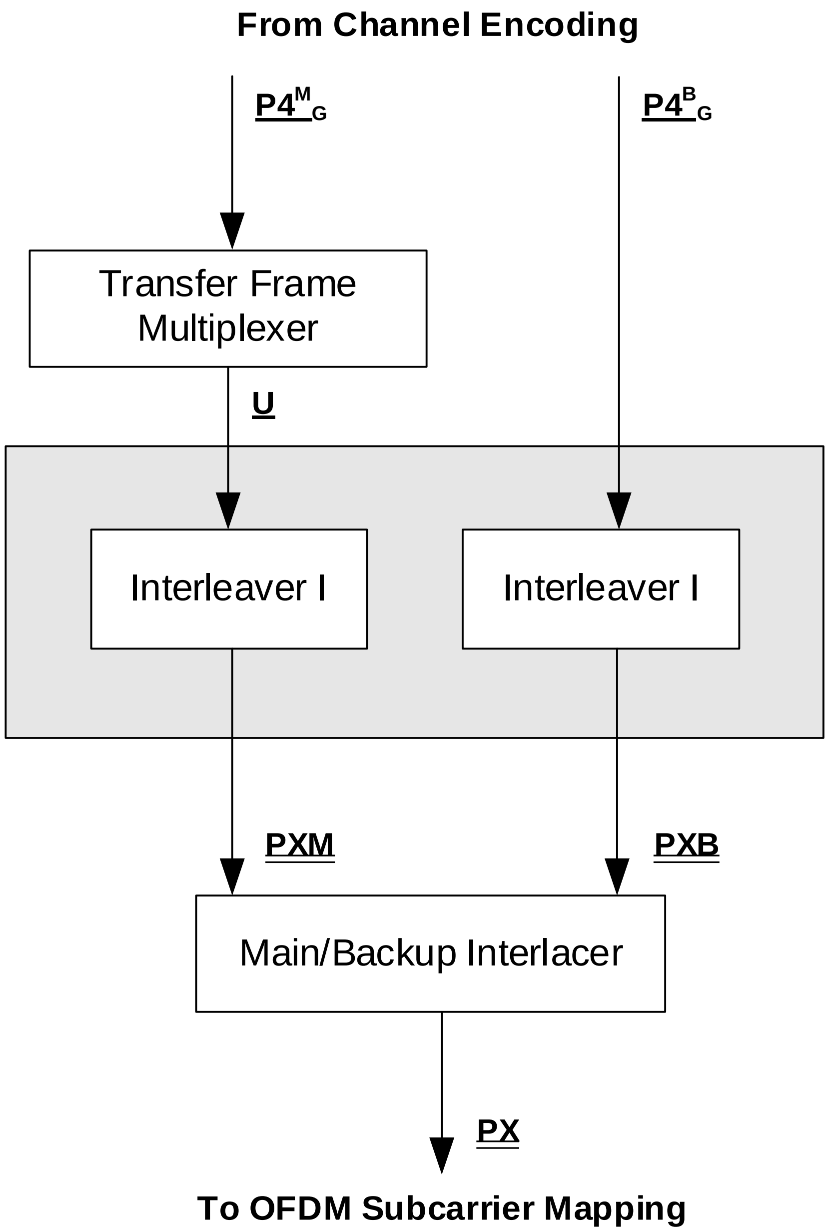


Figure 10-10: PX IP – Service Modes MP1X and MP1XOV

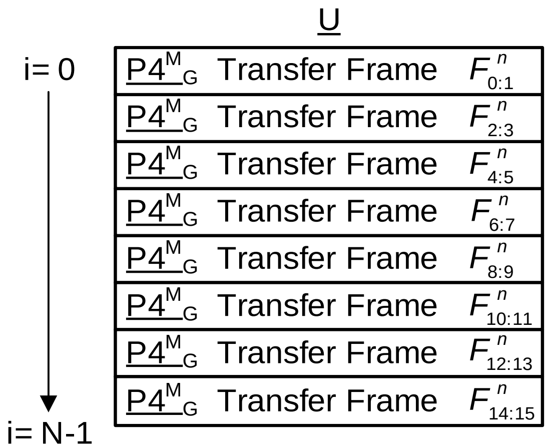


Figure 10-11: PX Transfer Frame Multiplexer Output – Service Modes MP1X and MP1XOV

Table 10-24: PXB Interleaver I Parameter Values – Service Modes MP1X and MP1XOV

Service Mode	J	B	C	M	$\underline{v}$	b	I_0	N
MP1X	16	2	18	1	See Note 1 below	N/A	N/A	18432
MP1XOV	8	2	18	1	See Note 2 below	N/A	N/A	9216

Note 1: *$\underline{v}$ = [14,10,3,7,13,4,8,1,15,11,2,6,12,5,9,0]*

Note 2: *$\underline{v}$ = [7,5,1,3,6,2,4,0]*

Table 10-25: PXM Interleaver I Parameter Values – Service Modes MP1X and MP1XOV

Service Mode	J	B	C	M	$\underline{v}$	b	I_0	N
MP1X	16	16	18	1	See Note 1 below	N/A	N/A	147456
MP1XOV	8	16	18	1	See Note 2 below	N/A	N/A	73728

Note 1: *$\underline{v}$ = [4,8,7,10,14,3,13,1,5,9,6,11,15,2,12,0]*

Note 2: *$\underline{v}$ = [2,4,3,5,7,1,6,0]*

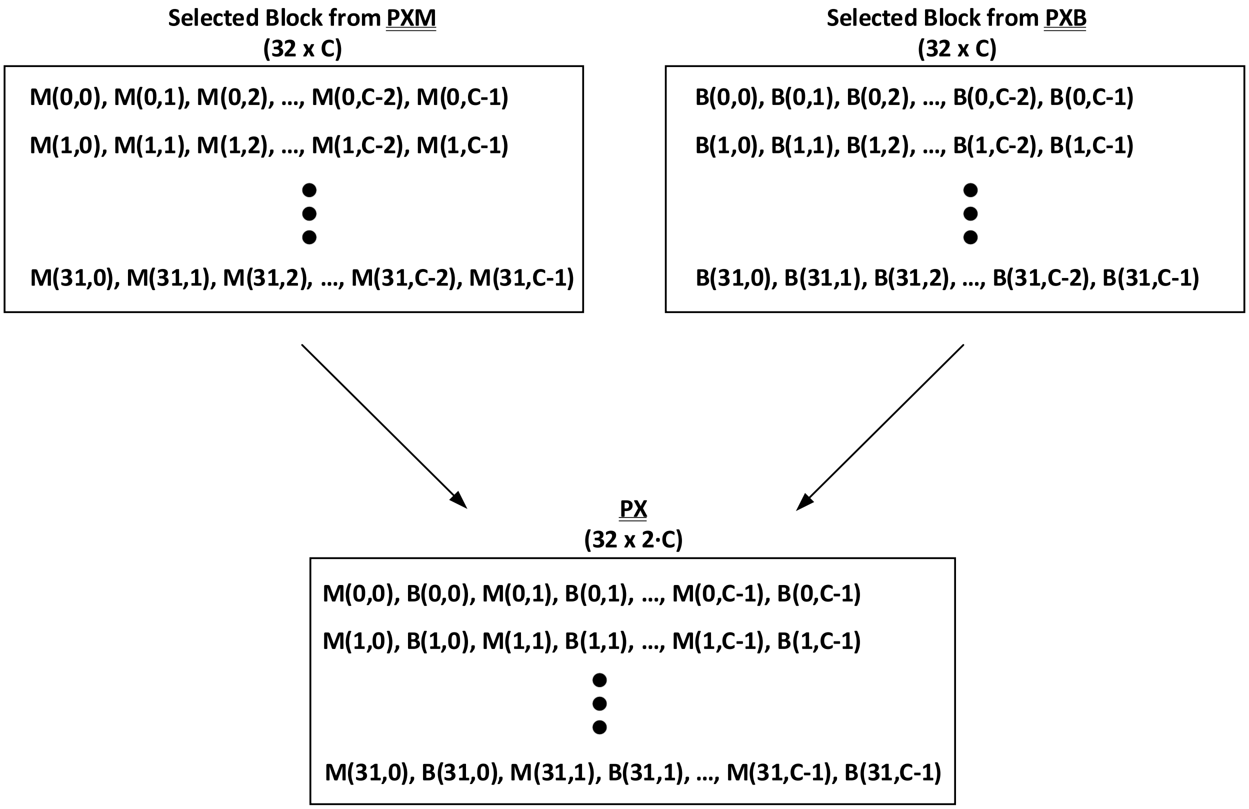


Figure 10-12: Main/Backup Interlacing Illustration for PX Interleaver Matrix

10.4.3 PB Interleaving Process

The PB IP interleaves the bits mapped to the Primary Main and Primary Extended sidebands depicted in Figure 7-9 through Figure 7-13. The PB IP disperses P1 and PIDS logical channels into the PB interleaver matrix in advanced primary service modes DSB1 and DSB1OV. The PB IP also disperses POV and PIDSOV logical channels into the PBOV interleaver matrix in advanced primary service modes MP1XOV, MP6OV, DSB1OV.

10.4.3.1 Service Mode DSB1

Figure 10-13 shows the PB IP for advanced service mode DSB1. In these service modes, the PB IP consists of a Type I interleaver and a Type II interleaver for processing $P1^B_G$ and $PIDS^B_G$ transfer frames, respectively, for output to the PBB interleaver matrix. It likewise contains Type I and Type II interleavers for processing $P1^M_G$ and $PIDS^M_G$ transfer frames for output to the PBM interleaver matrix. The number of transfer frames required to fill the PBM and PBB interleaver matrices in service mode DSB1 is shown in Table 10-8. The PBB and PBM interleaver matrices are subsequently interlaced into a single output interleaver matrix, PB.

The $P1^B_G$ and $P1^M_G$ transfer frames arrive from channel encoding at the block-pair rate Rp. The $P1^B_G$ transfer frames are interleaved into the PBB interleaver matrix immediately upon receipt at each block-pair boundary, completely filling the interleaver with a single transfer frame. However, since $P1^M_G$ is interleaved over an entire L1 frame, the transfer frame multiplexer collects and concatenates eight transfer frames into a single vector U prior to interleaving on the L1 frame boundary, as shown in Figure 10-14. The first bit of the first transfer frame becomes the first bit of U. The first bit of each subsequent transfer frame follows the last bit of the previous transfer frame.

The $\underline{\underline{PBM}}$ and $\underline{\underline{PBB}}$ interleaver matrices are processed in parallel, as shown in Figure 10-13. The steps required to process the $\underline{\underline{PBM}}$ interleaver inputs for L1 frame n are given as follows:

1. ***Multiplex $\underline{P1^M_G}$ transfer frames***

   The $\underline{P1^M_G}$ transfer frames are multiplexed into the vector $\underline{U}$ as shown in Figure 10-14.

2. ***Interleave multiplexer output***

   The vector $\underline{U}$ is interleaved into $\underline{\underline{PBM}}$ using Interleaver I with the parameters shown in Table 10-26 for service mode DSB1. The vector $\underline{U}$ is dispersed over the full row and column span of $\underline{\underline{PBM}}$, leaving holes to be filled by Interleaver II with $\underline{PIDS^M_G}$ data.

Table 10-26: PBM Interleaver I Parameter Values for Service Mode DSB1

Service Mode	J	B	C	M	$\underline{v}$	b	I_0	N
DSB1	56	16	18	1	See Note 1 below	N/A	N/A	508928

Note 1: *$\underline{v}$ = [22,4,49,18,28,40,8,26,0,53,39,32,45,14,21,7,50,17,29,42,10,25,2,52,37,34, 46,12,23,5,51,19,31,41,11,27,1,54,36,33,44,13,20,6,48,16,30,43,9,24,3,55,38,35,47,15]*

3. ***Interleave $\underline{PIDS^M_G}$ transfer frames***

   Each $\underline{PIDS^M_G}$ transfer frame is interleaved into $\underline{\underline{PBM}}$ using Interleaver II with the parameters shown in Table 10-27 for service mode DSB1.

Table 10-27: PBM Interleaver II Parameter Values for Service Mode DSB1

Service Mode	J	B	C	M	v	b	I_0	N
DSB1	56	16	18	1	See Note 1 below	448	508928	7168

Note 1: v = [*22,4,49,18,28,40,8,26,0,53,39,32,45,14,21,7,50,17,29,42,10,25,2,52,37,34, 46,12,23,5,51,19,31,41,11,27,1,54,36,33,44,13,20,6,48,16,30,43,9,24,3,55,38,35,47,15*]

Interleaver II constrains the row span of each interleaved $\underline{PIDS}^{M}_{G}$ transfer frame to one interleaver block (32 rows). This is accomplished by properly setting the interleaver variable i before each execution of Interleaver II. Table 10-28 shows the relationship between the variable i and the BC of the $\underline{PIDS}^{M}_{G}$ transfer frames in service mode DSB1.

Table 10-28: Bit Numbering of $\underline{PIDS}^{M}_{G}$ Transfer Frames in Service Mode DSB1

BC	DSB1 Range of variable i
0	448…895
1	896…1343
2	1344…1791
3	1792…2239
4	2240…2687
5	2688…3135
6	3136…3583
7	3584…4031
8	4032…4479
9	4480…4927
10	4928…5375
11	5376…5823
12	5824…6271
13	6272…6719
14	6720…7167
15	0…447

When Interleaver I has processed one vector U and Interleaver II has processed one $\underline{PIDS}^{M}_{G}$ transfer frame, a 32 x J *submatrix* of PBM is completely full and ready for OFDM Subcarrier Mapping. Each successive $\underline{PIDS}^{M}_{G}$ transfer frame is interleaved over the next successive interleaver block (for J interleaver partitions). After each $\underline{PIDS}^{M}_{G}$ transfer frame is processed by Interleaver II, the next 32 x J submatrix of PBM is available for OFDM Subcarrier Mapping.

After Interleaver I has processed one vector U and Interleaver II has processed sixteen $\underline{PIDS}^{M}_{G}$ transfer frames, interleaver matrix PBM is completely filled and the processing flow resets.

The PBB interleaver matrix has a span of only two blocks (as compared to the sixteen-block PBM interleaver matrix). The steps required to process the PBB interleaver inputs for a single block pair in L1 frame n are given as follows:

1. ***Interleave $\underline{P1}^{B}_{G}$ transfer frame***

A $P1^B_G$ transfer frame is interleaved into PBB using Interleaver I with the parameters shown in Table 10-29 for service mode DSB1. $P1^B_G$ is dispersed over the full row and column span of PBB, leaving holes to be filled by Interleaver II with $PIDS^B_G$ data.

Table 10-29: PBB Interleaver I Parameter Values for Service Mode DSB1

Service Mode	J	B	C	M	v	b	I_0	N
DSB1	56	2	18	1	See Note 1 below	N/A	N/A	63616

Note 1: *v = [40,32,49,45,26,4,14,39,28,53,18,22,0,8,42,34,50,46,25,7,12,37,29,52,17,21, 2,10,41,33,51,44,27,5,13,36,31,54,19,23,1,11,43,35,48,47,24,6,15,38,30,55,16,20,3,9]*

2. ***Interleave $PIDS^B_G$ transfer frames***

Each $PIDS^B_G$ transfer frame is interleaved into PBB using Interleaver II with the parameters shown in Table 10-30 for service mode DSB1.

Table 10-30: PBB Interleaver II Parameter Values for Service Mode DSB1

Service Mode	J	B	C	M	v	b	I_0	N
DSB1	56	2	18	1	See Note 1 below	448	63616	896

Note 1: v = [*40,32,49,45,26,4,14,39,28,53,18,22,0,8,42,34,50,46,25,7,12,37,29,52,17,21, 2,10,41,33,51,44,27,5,13,36,31,54,19,23,1,11,43,35,48,47,24,6,15,38,30,55,16,20,3,9*]

Interleaver II constrains the row span of each interleaved $PIDS^B_G$ transfer frame to one interleaver block (32 rows). This is accomplished by properly setting the interleaver variable i before each execution of Interleaver II. Table 10-31 shows the relationship between the variable i and the BC of the $PIDS^B_G$ transfer frames in service mode DSB1. Note that the range of i repeats every two blocks (the depth of the PBB interleaver).

Table 10-31: Bit Numbering of $PIDS^B_G$ Transfer Frames in Service Mode DSB1

BC	DSB1 Range of variable i
0	448…895
1	0…447
2	448…895
3	0…447
4	448…895
5	0…447
6	448…895
7	0…447
8	448…895
9	0…447
10	448…895
11	0…447
12	448…895
13	0…447

BC	DSB1 Range of variable i
14	448…895
15	0…447

When Interleaver I has processed one $P1^B_G$ transfer frame and Interleaver II has processed one $PIDS^B_G$ transfer frame, a 32 x J submatrix of PBB is completely full and ready for OFDM Subcarrier Mapping. Each successive $PIDS^B_G$ transfer frame is interleaved over the next successive interleaver block (for J interleaver partitions). After each $PIDS^B_G$ transfer frame is processed by Interleaver II, the next 32 x J submatrix of PBB is available for OFDM Subcarrier Mapping.

After Interleaver I has processed one $P1^B_G$ transfer frame and Interleaver II has processed two $PIDS^B_G$ transfer frames, interleaver matrix PBB is completely filled and the processing flow resets.

After interleaving, bits from a single block of current active interleaver matrices PBB and PBM are interlaced to form the PB interleaver matrix. The PBB interleaver matrix has a depth of two blocks, while PBM has a depth of sixteen blocks. For a given block *B* (ranging from 0 to 15), bits are selected from block *B modulo 2* of the current instance of PBB, and bits are selected from block *B* of the current instance of PBM.

Interlacing consists of alternately extracting bits, row by row, from the selected block of PBM and PBB and writing them into a single PB interleaver matrix. Since there are *C* columns in PBB and PBM, PB has dimensions of 32 rows by 2·C columns. Thus, for a given row i (ranging from 0 to 31):

$$PB(i) = \{PBM(i,0), PBB(i,0), PBM(i,1), PBB(i,1), \ldots, PBM(i,C-1), PBB(i,C-1)\}$$

Main/backup interlacing is illustrated in Figure 10-15.

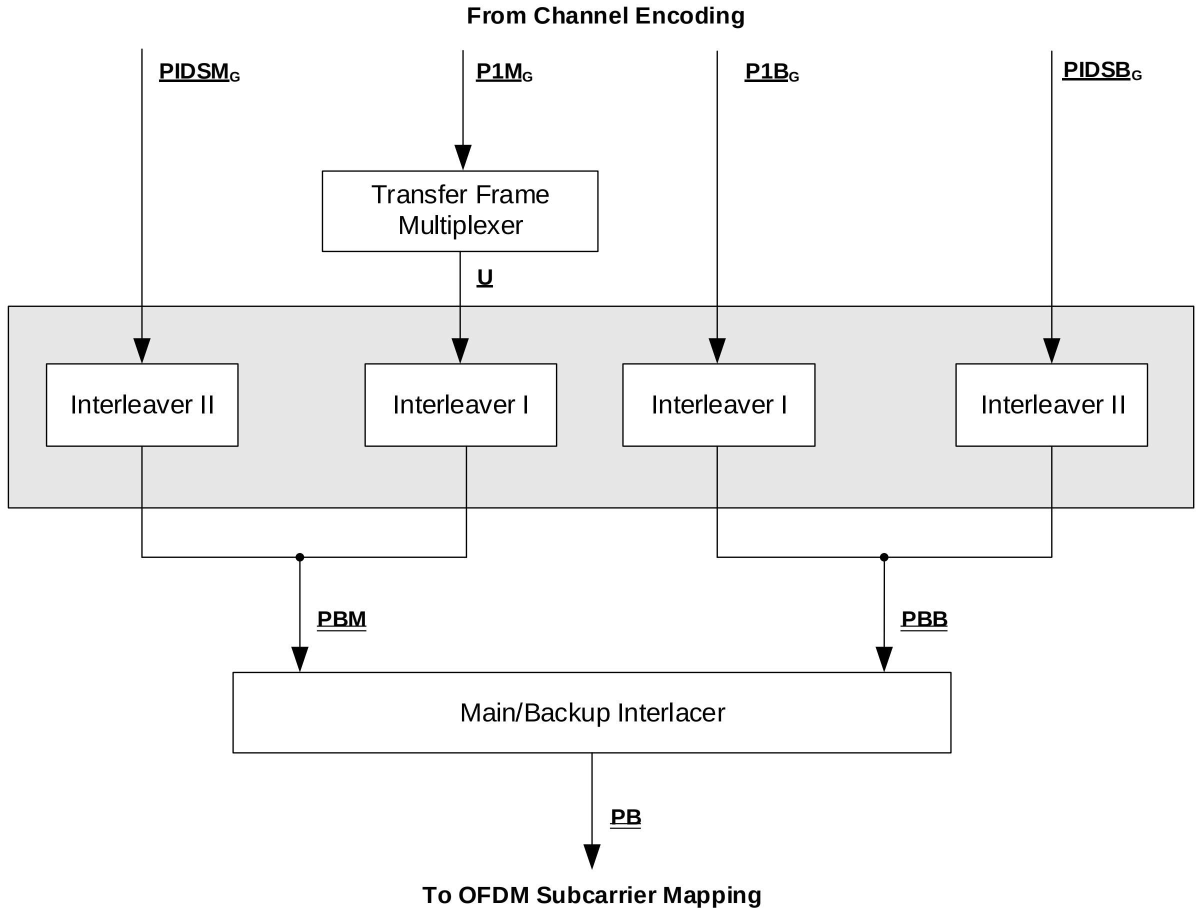


Figure 10-13: PB IP – Service Mode DSB1

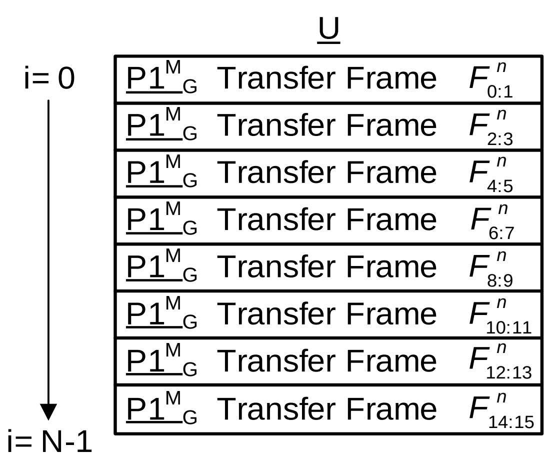


Figure 10-14: PB Transfer Frame Multiplexer Output – Service Mode DSB1

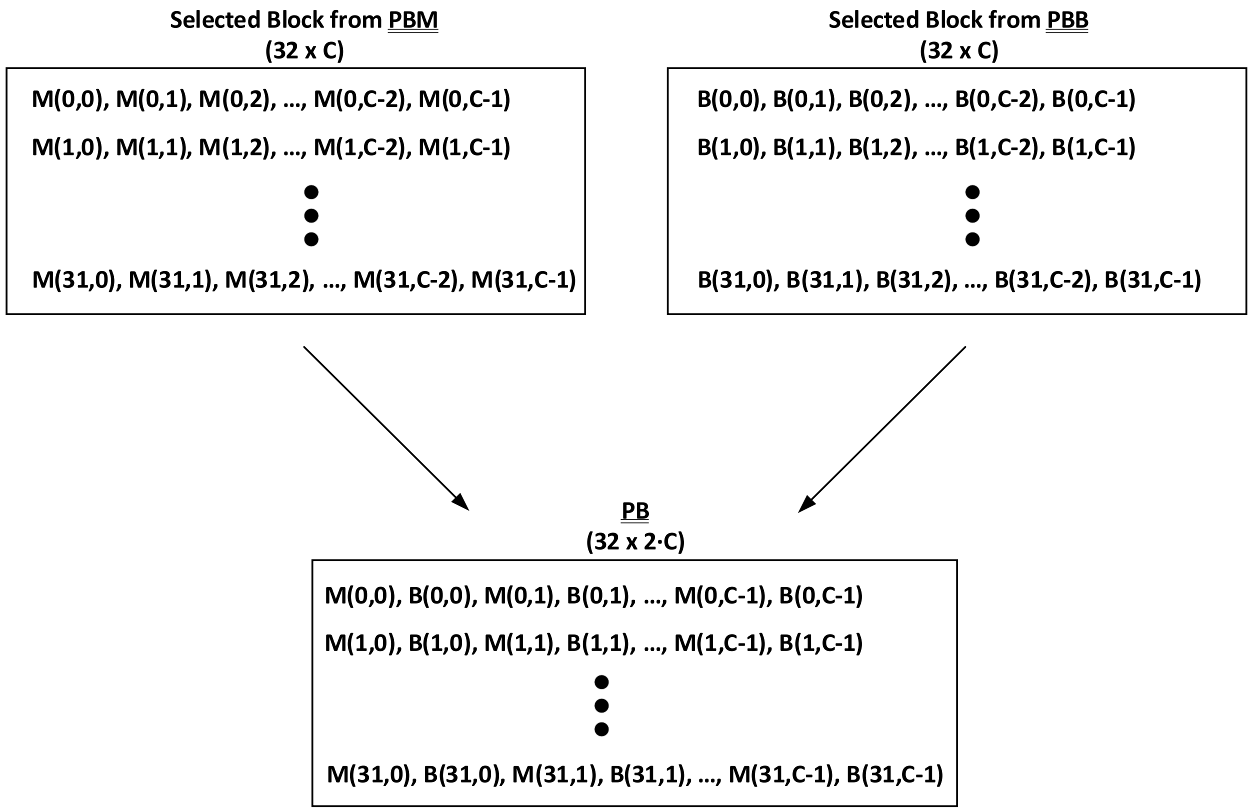


Figure 10-15: Main/Backup Interlacing Illustration for PB Interleaver Matrix

10.4.3.2 Service Modes MP1XOV and MP6OV

Figure 10-16 shows the PB IP for advanced service modes MP1XOV and MP6OV. The processing in MP1XOV and MP6OV is identical to that of DSB1, except for naming conventions: the P1 and PIDS logical channels are renamed the POV and PIDSOV logical channels, respectively, and the PB interleaver matrix is renamed the PBOV interleaver matrix.

Besides these terminology differences, all processing details are identical to those described in Subsection 10.4.3.1 for service mode DSB1.

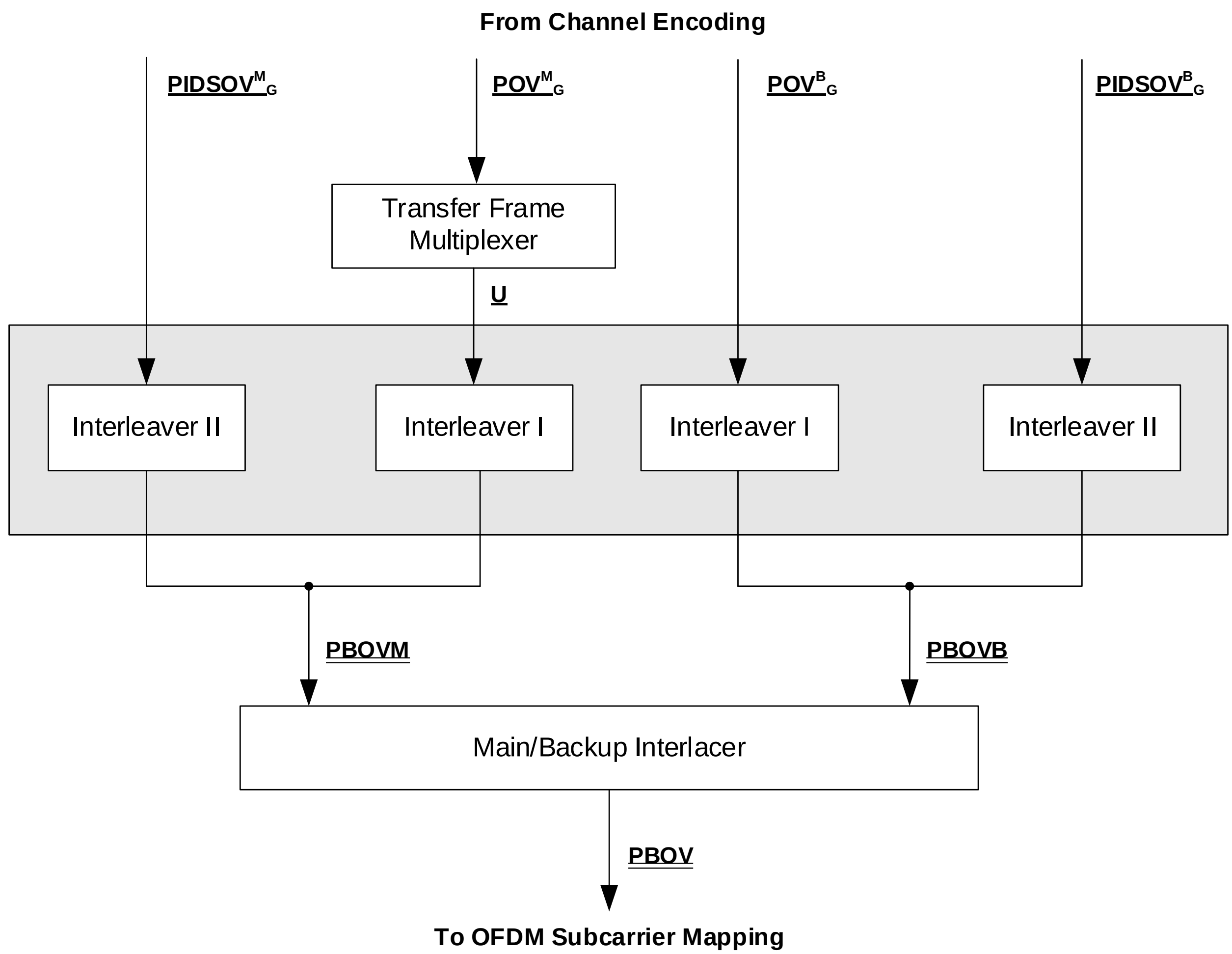


Figure 10-16: PB IP – Service Modes MP1XOV and MP6OV

10.4.3.3 Service Mode DSB1OV

Figure 10-17 shows the PB IP for advanced service mode DSB1OV. In service mode DSB1OV, the P1 and PIDS processing of DSB1 and the POV and PIDSOV processing of MP1XOV (or MP6OV) are performed in parallel to create PB and PBOV interleaver matrices, respectively.

All processing details are identical to those described in Subsection 10.4.3.1 for DSB1 and Subsection 10.4.3.2.

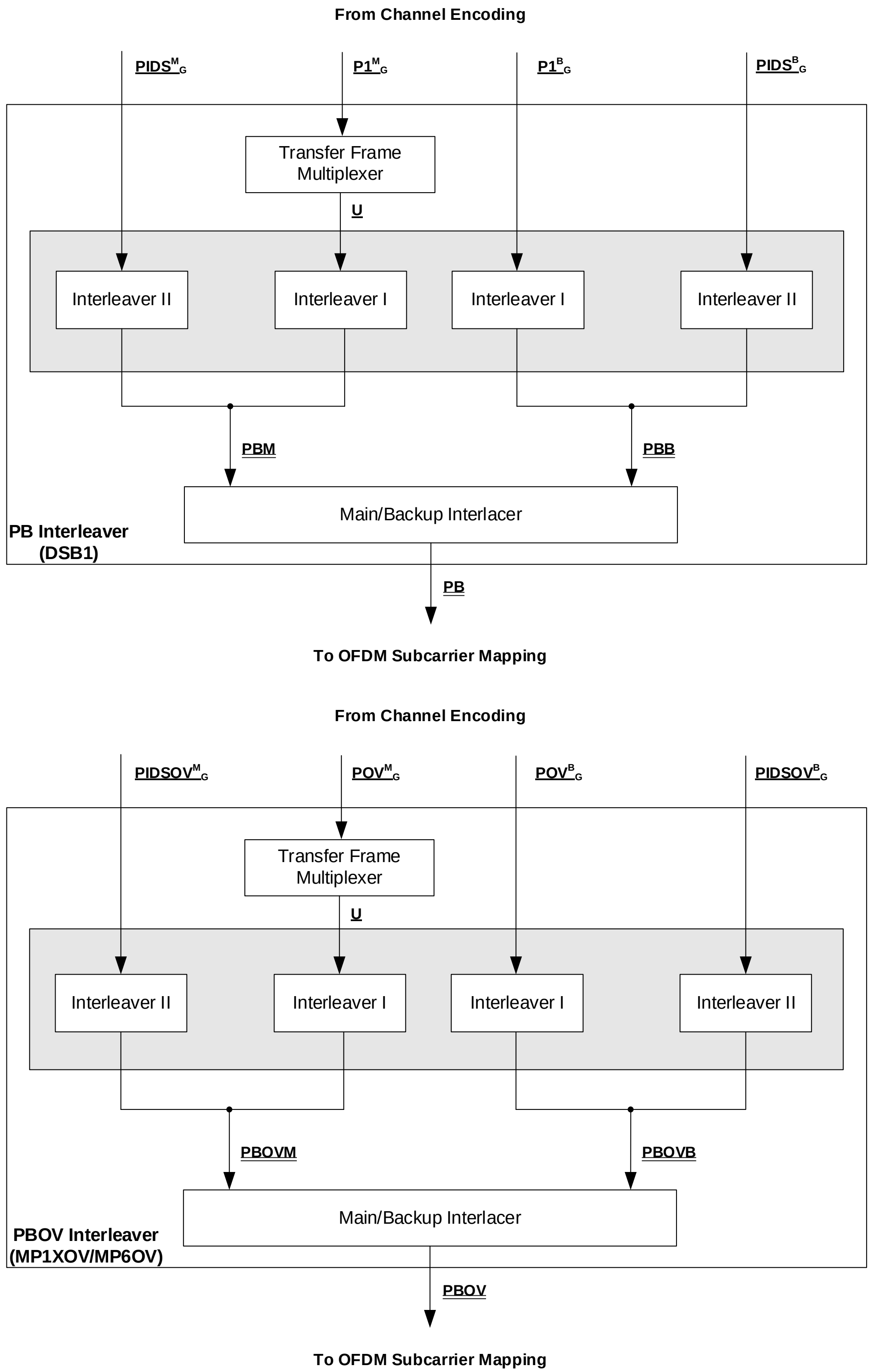


Figure 10-17: PB IP – PB and PBOV Interleaver Matrices in Service Mode DSB1OV

10.4.4 SB Interleaving Process

The SB IP interleaves bits destined for the upper and lower secondary sidebands depicted in Figure 7-15. This IP is active in secondary service mode MS5 for use in All Digital waveforms.

Figure 10-18 shows the SB IP for advanced secondary service mode MS5. The processing in MS5 is identical to that of DSB1, except for naming conventions: the P1 and PIDS logical channels are renamed the S1 and SIDS logical channels, respectively, and the PB interleaver matrix is renamed the SB interleaver matrix.

Besides these terminology differences, all processing details are identical to those described in Subsection 10.4.3.1 for service mode DSB1.

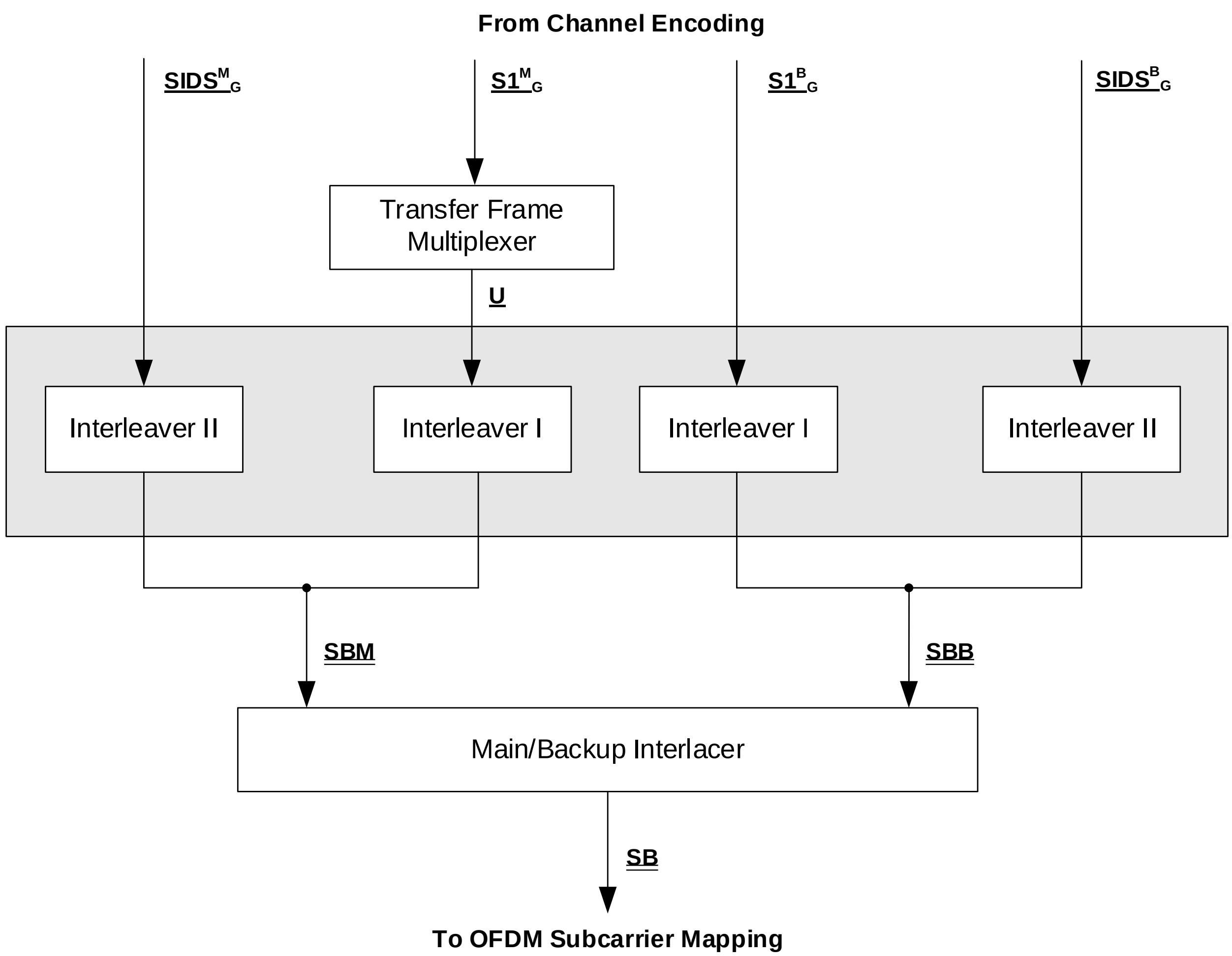


Figure 10-18: SB IP – Service Mode MS5

11 System Control Processing

11.1 Introduction

Under the direction of the upper layers, System Control Processing assembles and differentially encodes a sequence of bits (system control data sequence) destined for each reference subcarrier, as shown in Figure 11-1. There are up to 61 reference subcarriers, numbered 0 through 60 (with reference subcarrier 30 unused), that are distributed throughout the OFDM spectrum (see Figure 5-3 and Figure 5-4). The number of reference subcarriers broadcast in a given waveform depends on the service mode; however, System Control Processing always outputs all 61 system control data sequences, regardless of service mode.

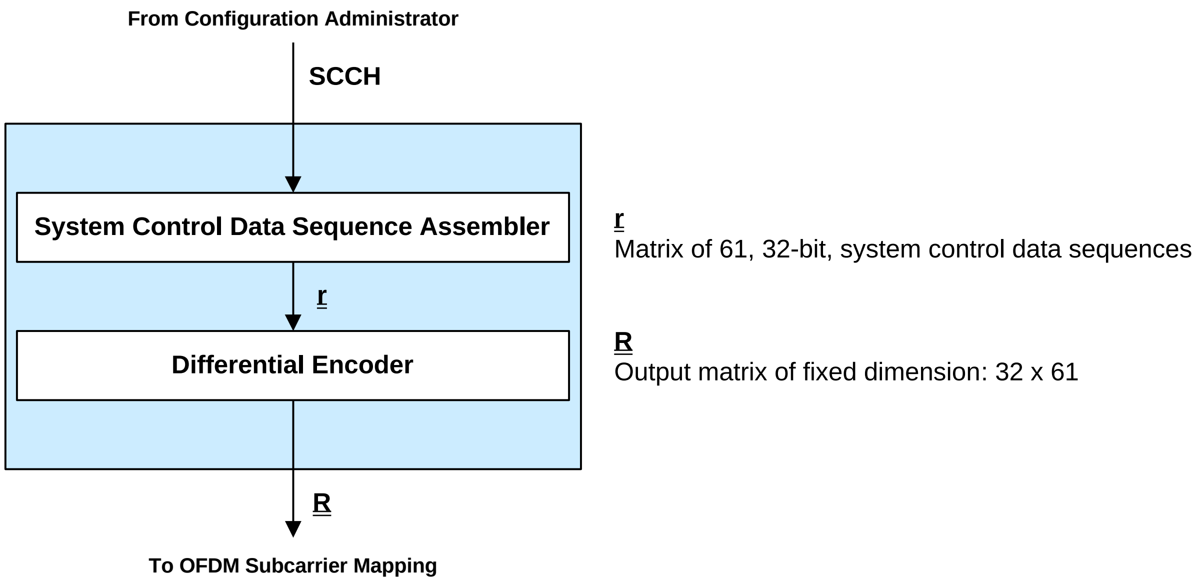


Figure 11-1: System Control Processing Conceptual Diagram

As shown in Figure 11-1, System Control Processing receives inputs from the Configuration Administrator via the SCCH. This system control is defined in Section 6 and is composed of primary, secondary, and advanced service mode control, plus primary and secondary reserved bits. The size and rate of this transfer is defined in Table 6-1.

Using the system control inputs, the System Control Data Sequence Assembler creates the system control bit sequence over T_b for each of the 61 reference subcarriers. This is a matrix $\underline{r}$ (lowercase) of 61, 32-bit, system control data sequences. The Differential Encoder then differentially encodes each bit sequence in time. The resulting output is a matrix $\underline{\underline{R}}$ (uppercase) of fixed dimension 32 x 61. The row dimension of $\underline{\underline{R}}$ corresponds to the number of OFDM symbols per T_b and the column dimension corresponds to the maximum number of active reference subcarriers per OFDM symbol.

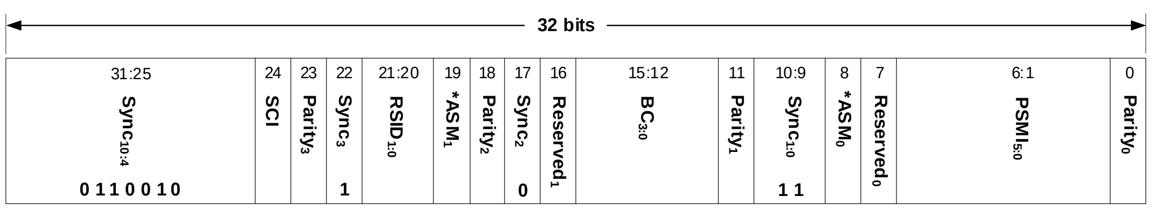
The matrix $\underline{\underline{R}}$ is available to OFDM Subcarrier Mapping at the rate R_b. In addition, System Control Processing provides the L1 block count to Layer 2 at the rate R_b via the SCCH.

11.2 System Control Data Sequence Assembler

The System Control Data Sequence Assembler collects all system control information from the Configuration Administrator and, together with some layer control information, develops a matrix $\underline{\underline{r}}$ of 61, 32-bit, system control data sequences. The rows of $\underline{\underline{r}}$ are numbered 0,1,2,3,…,31 and the columns are numbered 0,1,2,3,…,60. Each row of $\underline{\underline{r}}$ contains one bit of the system control data sequence for each reference subcarrier (before *differential encoding*) and is transmitted in the same OFDM symbol. Row 0 is populated first. Any given column of $\underline{\underline{r}}$ contains the system control data sequence for a single reference subcarrier over 32 OFDM symbols.

The system control data sequence consists of bit fields that represent the various system control components. Reference subcarriers located in primary sidebands have different fields than reference subcarriers located in secondary sidebands. Information in the primary reference subcarriers applies only to primary services and information in the secondary reference subcarriers applies only to secondary services. Refer to Table 12-24 for the column indices of $\underline{\underline{R}}$ that map to primary reference subcarriers; refer to Table 12-25 for the column indices of $\underline{\underline{R}}$ that map to secondary reference subcarriers.

The primary reference subcarrier system control data sequence is depicted in Figure 11-2 and defined in Table 11-1. Bits 31 through 0 map to rows 0 through 31 of $\underline{\underline{r}}$, respectively.



$*ASM_0 = 1$ and $ASM_1 = 0$ if $PSMI_{5:0} \neq 000000$

Figure 11-2: Primary Reference Subcarrier System Control Data Sequence

Table 11-1: Primary System Control Data Sequence Bit Map

Field	Bit Index	Bit Length	Description
$Sync_{10:4}$	31:25	7	$Sync_{10:4}$ = 0110010
Secondary Channel Indicator (SCI)	24	1	0 = primary only (Hybrid or Extended Hybrid) 1 = primary and secondary (All Digital)
$Parity_3$	23	1	Even parity for SCI
$Sync_3$	22	1	$Sync_3$ = 1
Reference Subcarrier Identification ($RSID_{1:0}$)	21:20	2	Fixed two-bit identifier per reference subcarrier
ASM_1	19	1	Advanced Service Mode Control bit 1. Unless PSMI = 0, must be set to 0 for backward compatibility with first-generation receivers
$Parity_2$	18	1	Even parity for ASM_1 and $RSID_{1:0}$

Field	Bit Index	Bit Length	Description
$Sync_2$	17	1	$Sync_2 = 0$
$Reserved_1$	16	1	Controlled by the Configuration Administrator
L1 Block Count ($BC_{3:0}$)	15:12	4	Modulo-16 count, which increments every 32 OFDM symbols
$Parity_1$	11	1	Even parity for $BC_{3:0}$ and $Reserved_1$
$Sync_{1:0}$	10:9	2	$Sync_{1:0} = 11$
ASM_0	8	1	Advanced Service Mode Control bit 0. Unless PSMI = 0, must be set to 1 for backward compatibility with first-generation receivers
$Reserved_0$	7	1	Controlled by the Configuration Administrator
Primary Service Mode Indicator ($PSMI_{5:0}$)	6:1	6	Primary service mode value
$Parity_0$	0	1	Even parity for $PSMI_{5:0}$, $Reserved_0$, and ASM_0

The secondary reference subcarrier system control data sequence is depicted in Figure 11-3 and defined in Table 11-2. Bits 31 through 0 map to rows 0 through 31 of r, respectively.

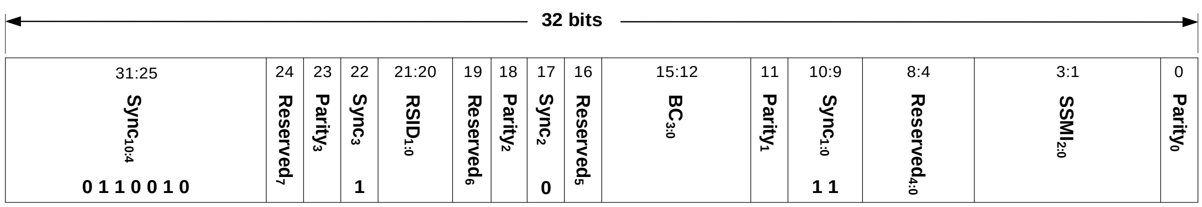


Figure 11-3: Secondary Reference Subcarrier System Control Data Sequence

NOTE | Figure 11-3 has been updated since NRSC-5-D

Table 11-2: Secondary System Control Data Sequence Bit Map

Field	Bit Index	Bit Length	Description
$Sync_{10:4}$	31:25	7	$Sync_{10:4}$ = 0110010
$Reserved_7$	24	1	Controlled by the Configuration Administrator
$Parity_3$	23	1	Even parity for $Reserved_7$
$Sync_3$	22	1	$Sync_3$ = 1
Reference Subcarrier Identification ($RSID_{1:0}$)	21:20	2	Fixed two-bit identifier per reference subcarrier
$Reserved_6$	19	1	Controlled by the Configuration Administrator
$Parity_2$	18	1	Even parity for $RSID_{1:0}$ and $Reserved_6$
$Sync_2$	17	1	$Sync_2$ = 0
$Reserved_5$	16	1	Controlled by the Configuration Administrator
L1 Block Count ($BC_{3:0}$)	15:12	4	Modulo-16 count, which increments every 32 OFDM symbols
$Parity_1$	11	1	Even parity for $Reserved_5$ and $BC_{3:0}$
$Sync_{1:0}$	10:9	2	$Sync_{1:0}$ = 11
$Reserved_{4:0}$	8:4	5	Controlled by the Configuration Administrator
Secondary Service Mode Indicator ($SSMI_{2:0}$)	3:1	3	Secondary service mode value
$Parity_0$	0	1	Even parity for $Reserved_{4:0}$ and $SSMI_{2:0}$

11.2.1 Block Synchronization

The sync bits serve to aid in receiver synchronization. The sync bit pattern is distributed over the system control data sequence as shown in Figure 11-2 and Figure 11-3.

11.2.2 Advanced Service Mode Control (ASM)

Some advanced service modes are not backward compatible with any standard service mode and are designated as primary service mode 0. First-generation receivers cannot acquire these signals because primary service mode 0 is not defined for them. When the primary service mode is not 0, ASM_0 must be set to 1 and ASM_1 must be set to 0 to preserve backward compatibility with first-generation receivers.

The two ASM bits specify which of the non-backward-compatible advanced service modes is configuring the transmitted waveform when the primary service mode is 0. The Advanced Service Mode Control bit mapping is defined in Table 6-4.

There are no ASM bits in the secondary system control data sequence; instead, reserved bits occupy these positions.

11.2.3 Reference Subcarrier Identification

The Reference Subcarrier Identification ($RSID_{1:0}$) is a two-bit value that is applied to each reference subcarrier across the OFDM spectrum. The reference subcarrier identification maps to the reference subcarriers (columns of $\underline{\underline{r}}$) as specified in Table 11-3. The Reference Subcarrier ID does not uniquely identify a subcarrier. One use of this parameter is to assist the receiver in frequency acquisition and tracking.

Table 11-3: Reference Subcarrier Identification

Column Number of $\underline{\underline{r}}$	$RSID_{1:0}$ (bits 21:20)		Column Number of $\underline{\underline{r}}$	$RSID_{1:0}$ (bits 21:20)		Column Number of $\underline{\underline{r}}$	$RSID_{1:0}$ (bits 21:20)		Column Number of $\underline{\underline{r}}$	$RSID_{1:0}$ (bits 21:20)	
	21	20		21	20		21	20		21	20
0	1	0	16	1	0	32	1	0	48	1	0
1	0	1	17	0	1	33	1	1	49	1	1
2	0	0	18	0	0	34	0	0	50	0	0
3	1	1	19	1	1	35	0	1	51	0	1
4	1	0	20	1	0	36	1	0	52	1	0
5	0	1	21	0	1	37	1	1	53	1	1
6	0	0	22	0	0	38	0	0	54	0	0
7	1	1	23	1	1	39	0	1	55	0	1
8	1	0	24	1	0	40	1	0	56	1	0
9	0	1	25	0	1	41	1	1	57	1	1
10	0	0	26	0	0	42	0	0	58	0	0
11	1	1	27	1	1	43	0	1	59	0	1
12	1	0	28	1	0	44	1	0	60	1	0
13	0	1	29	0	1	45	1	1			
14	0	0	30	0	0	46	0	0			
15	1	1	31	0	1	47	0	1			

11.2.4 Secondary Channel Indicator

Since the secondary sidebands are not transmitted in all waveforms, the primary reference subcarriers must indicate their presence or absence. The Secondary Channel Indicator (SCI) is a single bit in the primary system control data sequence. It is set to 1 when the signal has secondary sidebands; otherwise, it is set to 0. There is no SCI bit in the secondary system control data sequence; instead, a reserved bit occupies this position.

11.2.5 L1 Block Count

The four-bit L1 block count ($BC_{3:0}$) is a modulo-16 count which increments every 32 OFDM symbols. The first L1 block count inserted into the system control data sequence is 0. The same value is applied to each of the 61 system control data sequences. The value of BC contained in the primary reference subcarrier system control data sequence is always the same as the BC contained in the secondary reference subcarrier system control data sequence. Refer to Subsection 6.4 for further definition.

The L1 block count bit map is shown in Table 11-4.

Table 11-4: L1 Block Count Bit Map

L1 Block Count	$BC_{3:0}$ Bit Assignment (bits 15 :12)			
	15	14	13	12
0	0	0	0	0
1	0	0	0	1
2	0	0	1	0
3	0	0	1	1
4	0	1	0	0
5	0	1	0	1
6	0	1	1	0
7	0	1	1	1
8	1	0	0	0
9	1	0	0	1
10	1	0	1	0
11	1	0	1	1
12	1	1	0	0
13	1	1	0	1
14	1	1	1	0
15	1	1	1	1

11.2.6 Primary Service Mode Indicator

The primary service mode in Layer 1, as defined in Section 6, conveys various combinations of digital audio and data. Six bits in the system control data sequence of the primary reference subcarriers have been allocated to the Primary Service Mode Indicator ($PSMI_{5:0}$), as defined in Table 6-2.

Each of the reserved primary service modes must maintain backward compatibility, as defined in Subsection 6.2.1. Thus, first generation receivers will configure themselves to one of the basic operational modes MP1, MP2, MP3, MP5, or MP6 when one of the reserved modes is detected, as shown in Table 6-5.

11.2.7 Secondary Service Mode

The secondary service mode in Layer 1, as defined in Section 6, conveys various combinations of digital audio and data. Three bits in the system control data sequence of the secondary reference subcarriers have been allocated to the Secondary Service Mode Indicator ($SSMI_{2:0}$), as defined in Table 6-3.

11.2.8 Reserved

The value of the reserved bits is determined by the Configuration Administrator as discussed in Section 6. The reserved bits remain the same during the duration of the L1 frame.

11.3 Differential Encoder

The bits in each column of the 32 x 61 matrix r, assembled by the System Control Data Sequence Assembler, are differentially encoded in accordance with Figure 11-4 and are output to the matrix R in the same order. Conceptually, this process can be viewed as 61 parallel differential encoders. For an individual differential encoder, the bits of a single column j of r are processed sequentially, from i = 0,1,2,3,…,31. One system control data sequence bit is input to a differential encoder at a time. This input bit is modulo-2 added with the previously stored output bit R[i-1][j] to form the latest output bit, R[i][j]. The resulting output bit stream will reverse polarity each time the input bit is a 1. The initial state of each differential encoder is 0.

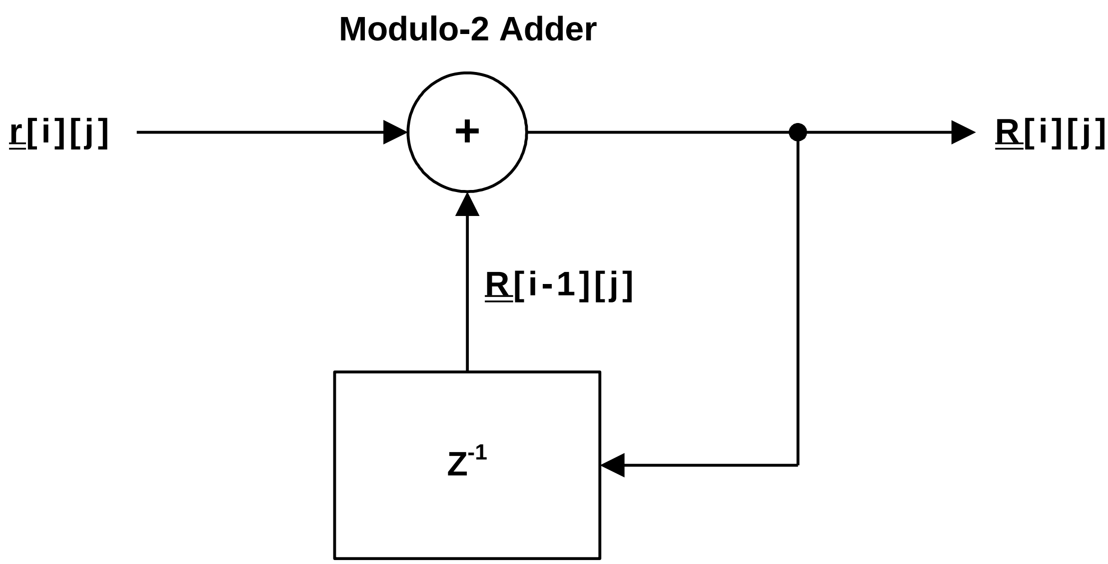


Figure 11-4: Differential Encoder

12 OFDM Subcarrier Mapping

12.1 Introduction

OFDM Subcarrier Mapping assigns interleaved bits to OFDM subcarriers. For each active interleaver matrix, OFDM Subcarrier Mapping assigns a row of bits from each interleaver partition to its respective frequency partition in the complex output vector $\underline{X}$. In addition, system control data sequence bits from a row of $\underline{\underline{R}}$ are mapped to the active reference subcarrier locations in $\underline{X}$. The service mode dictates which interleaver matrices and which elements of $\underline{\underline{R}}$ are active. Figure 12-1 shows the inputs, output, and component functions of OFDM Subcarrier Mapping.

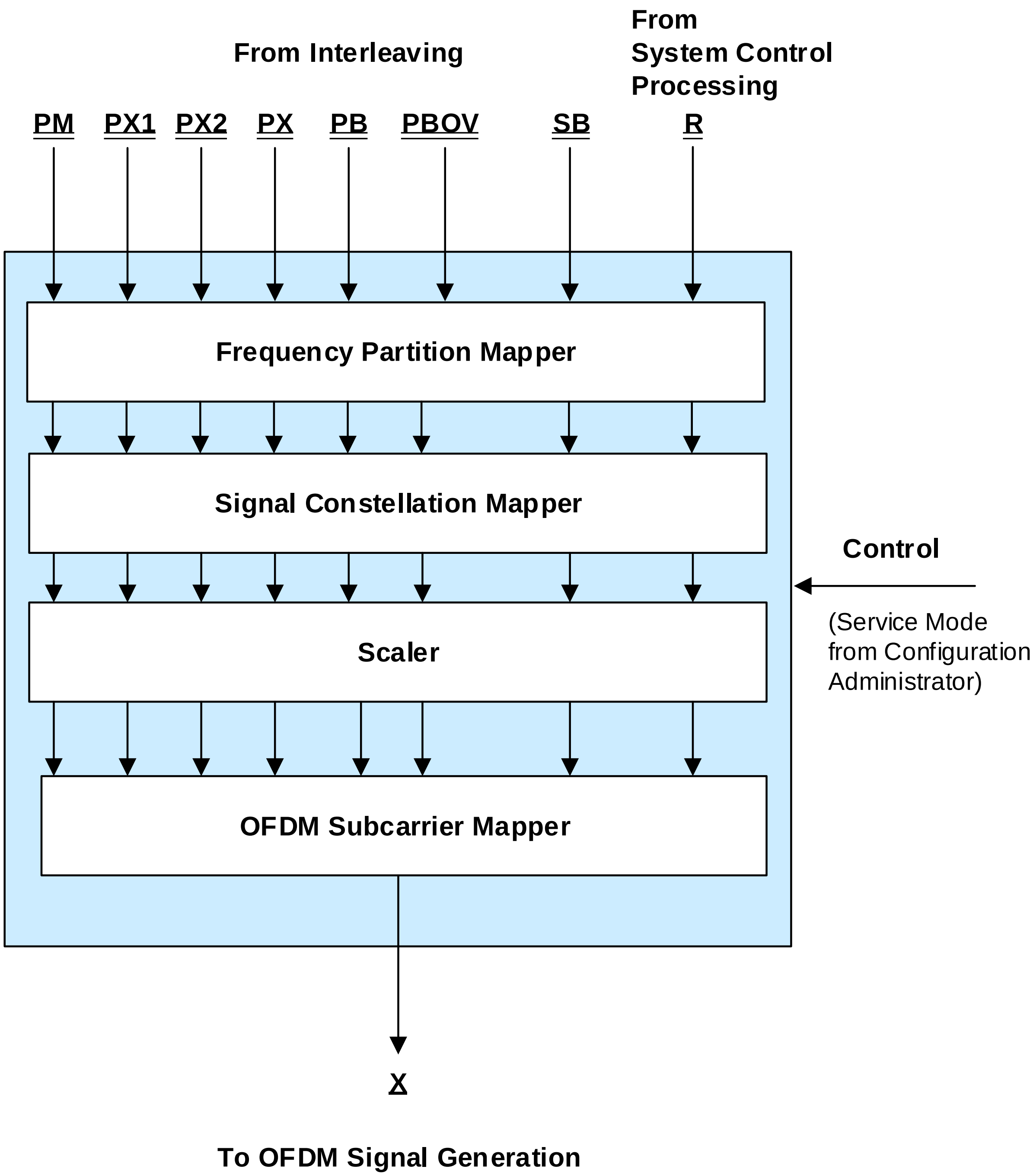


Figure 12-1: OFDM Subcarrier Mapping Conceptual Block Diagram

The inputs to OFDM Subcarrier Mapping are a row of bits from each active interleaver matrix and a row of bits from R, the matrix of system control data sequences.

The output from OFDM Subcarrier Mapping for each OFDM symbol is a single complex vector, X, of length 1093. The vector is indexed from k = 0,1,2,…,1092. The k^{th} element of X corresponds to subcarrier (k – 546), as shown in Figure 12-2.

Index into X	0	1	2	▪ ▪ ▪	1090	1091	1092
Subcarrier Number	-546	-545	-544		544	545	546

Figure 12-2: Assignment of Elements of Output Vector X to Subcarriers

Active elements in a row of R and the associated row from each active interleaver matrix are assigned to the same instance of X.

The *Frequency Partition Mapper* assigns bits from interleaver partitions to frequency partitions. For standard service modes, this is a trivial 1:1 mapping because QPSK modulation is applied to all OFDM data subcarriers. However, for those frequency partitions in advanced service modes in which 16-QAM or 64-QAM is used, there are two or three interleaver partitions, respectively, mapped to each frequency partition.

Specifically, for QPSK, one I bit and an adjacent Q bit from a single interleaver partition is assigned to each data subcarrier within a particular frequency partition. For 16-QAM, one I bit and an adjacent Q bit from one constituent interleaver partition, and another I bit and adjacent Q bit from a second constituent interleaver partition, are assigned to each data subcarrier within a particular frequency partition. In the same manner, for 64-QAM, three I/Q bit pairs from three different interleaver partitions are assigned to each data subcarrier within a particular frequency partition. This effectively increases data throughput for higher-order modulation schemes.

For example, in 16-QAM advanced service mode MP1X, data subcarriers in frequency partition 10 are assigned I/Q bit pairs from both PX interleaver partition 4 and PX interleaver partition 0. This process is described in detail in Subsection 12.2.1.1.

The *Signal Constellation Mapper* translates sets of bits read from frequency partitions and individual bits read from R to complex constellation values. The Scaler function applies the appropriate modulation and amplitude scale factors to these complex values. The OFDM Subcarrier Mapper then maps the scaled complex constellation values to the appropriate elements of the output vector X. Elements of X corresponding to unused subcarriers are set to the complex value 0 + *j*0.

12.2 Frequency Partition Mapper, Signal Constellation Mapper, and Scaler

The OFDM Subcarrier Mapping procedures for data and reference subcarriers are specified in Subsection 12.2.1 and Subsection 12.2.2, respectively.

12.2.1 Data Subcarrier Mapping Procedures

12.2.1.1 Frequency Partition Mapper

The mapping of interleaver partition to frequency partition is defined for each service mode in Table 12-1 through Table 12-14. Each table lists the service mode and its associated interleaver matrices, as well as their respective *modulation components*. The modulation component indicates the relative position of constituent I and Q bit pairs within indices to the complex constellation values assigned to the data subcarriers. For a given frequency partition, each modulation component is assigned bits from a single interleaver partition, as specified in the tables.

Within each interleaver partition, bits are ordered as I,Q. QPSK indices have just one modulation component, since QPSK modulates each data subcarrier with a single I and Q bit pair. However, 16-QAM indices require two modulation components – one for the most significant I and Q bits (MSB), and one for the least significant I and Q bits (LSB). Finally, 64-QAM indices need three modulation components – one for the most significant I and Q bits, one for the middle significant I and Q bits (XSB), and one for the least significant I and Q bits.

Advanced service modes MP1XOV, MP6OV, DSB1OV utilize *layered*, or *hierarchical*, *modulation*. For these service modes, one interleaver partition is assigned to the MSB of 16-QAM waveforms, two interleaver partitions are assigned to the MSB and XSB (middle significant bits) of 64-QAM waveforms, and a single interleaver partition from the PBOV interleaver matrix modulates the LSB.

In the tables, frequency partitions are arranged in order from lowest frequency to highest: lower primary sideband frequency partition 0 is farthest from channel center, while partition 13 is nearest to channel center. Likewise, upper primary sideband frequency partition 14 is closest to channel center, while partition 27 is farthest from channel center. In service mode MS5, lower secondary sideband frequency partition 28 is farthest from channel center, while partition 41 is nearest to channel center, and upper secondary sideband frequency partition 42 is closest to channel center, while partition 55 is farthest from channel center.

The unshaded numbers in the tables represent the interleaver partitions that are assigned to the indicated frequency partition.

Table 12-1: Frequency Partition Mapping for Service Mode MP1

Service Mode & Interleaver	Modulation Component	Lower Sideband Frequency Partition													
		0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**
MP1 PM	QPSK	0	1	2	3	4	5	6	7	8	9				
		Upper Sideband Frequency Partition													
		14	**15**	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**	**26**	**27**
						10	11	12	13	14	15	16	17	18	19

Table 12-2: Frequency Partition Mapping for Service Mode MP2

Service Mode & Interleaver	Modulation Component	Lower Sideband Frequency Partition													
		0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**
MP2 PM	QPSK	0	1	2	3	4	5	6	7	8	9				
MP2 PX1												0			
		Upper Sideband Frequency Partition													
		14	**15**	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**	**26**	**27**
MP2 PM						10	11	12	13	14	15	16	17	18	19
MP2 PX1					1										

Table 12-3: Frequency Partition Mapping for Service Mode MP3

Service Mode & Interleaver	Modulation Component	Lower Sideband Frequency Partition													
		0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**
MP3 PM	QPSK	0	1	2	3	4	5	6	7	8	9				
MP3 PX1												0	1		
		Upper Sideband Frequency Partition													
		14	**15**	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**	**26**	**27**
MP3 PM						10	11	12	13	14	15	16	17	18	19
MP3 PX1				2	3										

Table 12-4: Frequency Partition Mapping for Service Mode MP11

Service Mode & Interleaver	Modulation Component	Lower Sideband Frequency Partition													
		0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**
MP11 PM	QPSK	0	1	2	3	4	5	6	7	8	9				
MP11 PX1												0	1		
MP11 PX2														0	1
		Upper Sideband Frequency Partition													
		14	**15**	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**	**26**	**27**
MP11 PM						10	11	12	13	14	15	16	17	18	19
MP11 PX1				2	3										
MP11 PX2		2	3												

Table 12-5: Frequency Partition Mapping for Service Mode MP5

Service Mode & Interleaver	Modulation Component	Lower Sideband Frequency Partition													
		0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**
MP5 PM	QPSK	0	1	2	3	4	5	6	7	8	9				
MP5 PX1												0	1		
MP5 PX2														0	1
		Upper Sideband Frequency Partition													
		14	**15**	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**	**26**	**27**
MP5 PM						10	11	12	13	14	15	16	17	18	19
MP5 PX1				2	3										
MP5 PX2		2	3												

Table 12-6: Frequency Partition Mapping for Service Mode MP6

Service Mode & Interleaver	Modulation Component	Lower Sideband Frequency Partition													
		0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**
MP6 PM	QPSK	0	1	2	3	4	5	6	7	8	9				
MP6 PX2												0	1	2	3
		Upper Sideband Frequency Partition													
		14	**15**	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**	**26**	**27**
MP6 PM	QPSK					10	11	12	13	14	15	16	17	18	19
MP6 PX2		4	5	6	7										

Table 12-7: Frequency Partition Mapping for Service Mode MP1X

Service Mode & Interleaver	Modulation Component	Lower Sideband Frequency Partition													
		0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**
MP1X PM	QPSK	0	1	2	3	4	5	6	7	8	9				
MP1X PX	16-QAM MSB											4	5	6	7
	16-QAM LSB											0	1	2	3
		Upper Sideband Frequency Partition													
		14	**15**	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**	**26**	**27**
MP1X PM	QPSK					10	11	12	13	14	15	16	17	18	19
MP1X PX	16-QAM MSB	8	9	10	11										
	16-QAM LSB	12	13	14	15										

Table 12-8: Frequency Partition Mapping for Service Mode DSB1

Service Mode & Interleaver	Modulation Component	Lower Sideband Frequency Partition													
		0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**
DSB1 PB	16-QAM MSB	14	15	16	17	18	19	20	21	22	23	24	25	26	27
	16-QAM LSB	0	1	2	3	4	5	6	7	8	9	10	11	12	13
		Upper Sideband Frequency Partition													
		14	**15**	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**	**26**	**27**
	16-QAM MSB	28	29	30	31	32	33	34	35	36	37	38	39	40	41
	16-QAM LSB	42	43	44	45	46	47	48	49	50	51	52	53	54	55

Table 12-9: Reserved

Table 12-10: Frequency Partition Mapping for Service Mode MP1XOV

Service Mode & Interleaver	Modulation Component	Lower Sideband Frequency Partition													
		0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**
MP1XOV PM	16-QAM MSB	0	1	2	3	4	5	6	7	8	9				
MP1XOV PX												0	1	2	3
MP1XOV PBOV	16-QAM LSB	0	1	2	3	4	5	6	7	8	9	10	11	12	13
		Upper Sideband Frequency Partition													
		14	**15**	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**	**26**	**27**
MP1XOV PM	16-QAM MSB					10	11	12	13	14	15	16	17	18	19
MP1XOV PX		4	5	6	7										
MP1XOV PBOV	16-QAM LSB	14	15	16	17	18	19	20	21	22	23	24	25	26	27

Table 12-11: Frequency Partition Mapping for Service Mode MP6OV

Service Mode & Interleaver	Modulation Component	Lower Sideband Frequency Partition													
		0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**
MP6OV PM	16-QAM MSB	0	1	2	3	4	5	6	7	8	9				
MP6OV PX2	16-QAM MSB											0	1	2	3
MP6OV PBOV	16-QAM LSB	0	1	2	3	4	5	6	7	8	9	10	11	12	13
		Upper Sideband Frequency Partition													
		14	**15**	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**	**26**	**27**
MP6OV PM	16-QAM MSB					10	11	12	13	14	15	16	17	18	19
MP6OV PX2	16-QAM MSB	4	5	6	7										
MP6OV PBOV	16-QAM LSB	14	15	16	17	18	19	20	21	22	23	24	25	26	27

Table 12-12: Frequency Partition Mapping for Service Mode DSB1OV

Service Mode & Interleaver	Modulation Component	Lower Sideband Frequency Partition													
		0	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**13**
DSB1OV PB	64-QAM MSB	14	15	16	17	18	19	20	21	22	23	24	25	26	27
DSB1OV PB	64-QAM XSB	0	1	2	3	4	5	6	7	8	9	10	11	12	13
DSB1OV PBOV	64-QAM LSB	0	1	2	3	4	5	6	7	8	9	10	11	12	13
		Upper Sideband Frequency Partition													
		14	**15**	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**	**26**	**27**
DSB1OV PB	64-QAM MSB	28	29	30	31	32	33	34	35	36	37	38	39	40	41
DSB1OV PB	64-QAM XSB	42	43	44	45	46	47	48	49	50	51	52	53	54	55
DSB1OV PBOV	64-QAM LSB	14	15	16	17	18	19	20	21	22	23	24	25	26	27

Table 12-13: Reserved

Table 12-14: Frequency Partition Mapping for Service Mode MS5

Service Mode & Interleaver	Modulation Component	Lower Sideband Frequency Partition													
		28	29	30	31	32	33	34	35	36	37	38	39	40	41
MS5 SB	QPSK	14	15	16	17	18	19	20	21	22	23	24	25	26	27
		Upper Sideband Frequency Partition													
		42	43	44	45	46	47	48	49	50	51	52	53	54	55
		0	1	2	3	4	5	6	7	8	9	10	11	12	13

12.2.1.2 Signal Constellation Mapper and Scaler

Using the appropriate frequency partition mapping from Table 12-1 through Table 12-14, for each frequency partition, one row of bits from each of N mapped interleaver partitions is processed every T_s to obtain N I/Q bit pairs for each data subcarrier. The value of N is one for frequency partitions using QPSK, two for those using 16-QAM, or three for those using 64-QAM.

For 16-QAM, some interleaver partitions contain LSB I/Q bits and some contain MSB I/Q bits. The LSB and MSB I/Q bit pairs are combined to form a four-bit index into a 16-QAM constellation mapping table. For 64-QAM, some interleaver partitions contain LSB I/Q bits, some contain XSB I/Q bits, and some contain MSB I/Q bits. The LSB, XSB, and MSB I/Q bit pairs are combined to form a six-bit index into a 64-QAM constellation mapping table.

Specifically, for each QPSK data subcarrier within a frequency partition, one I bit and one Q bit from the assigned interleaver partition for a single specified modulation component are used to index one of four possible complex constellation values, as defined in Table 12-15. For 16-QAM waveforms, one I bit and one Q bit from the assigned interleaver partitions for the MSB and LSB modulation components index one of sixteen possible complex constellation values modulating each data subcarrier within a frequency partition, as defined in Table 12-16. Finally, for 64-QAM waveforms, one I bit and one Q bit from the assigned interleaver partitions for the MSB, XSB, and LSB modulation components index one of sixty-four possible complex constellation values modulating each data subcarrier within a frequency partition, as defined in Table 12-17.

Rows are processed sequentially, starting with the first row (row 0). When all rows of an interleaver matrix have been processed, the next instance of that interleaver matrix is processed, starting with the first row.

For a given row, bits are processed by frequency partition. Pairs of adjacent columns in each modulation component within a frequency partition are mapped to individual, complex, QAM data subcarriers. This mapping proceeds sequentially. The first two columns (0 and 1) are mapped to the starting subcarrier number of a frequency partition and the last two columns (34 and 35) are mapped to the ending subcarrier number of a frequency partition. Table 12-19 through Table 12-23 show the mapping to OFDM subcarrier numbers for each frequency partition in the waveform.

To map each adjacent *column pair* within each modulation component of a frequency partition to a subcarrier location within the vector $\underline{X}$, the following steps are taken:

1. Read a pair of bits from adjacent columns of each modulation component within a frequencypartition, starting with columns 0 and 1. For a given column pair, the bit read from the lower indexed column is mapped as an I bit, and the bit read from the higher indexed column is mapped as a Q bit.

2. Map the I/Q bit pair(s) from Step 1 to a complex constellation value using Table 12-15 for QPSK, Table 12-16 for 16-QAM, or Table 12-17 for 64-QAM. The I bit maps to the real component and the Q bit maps to the imaginary component of the constellation value.
3. Scale the I and Q components of the complex constellation value from Step 2 using the appropriate amplitude scale factor from Table 5-1 through Table 5-3. The amplitude scale factor is chosen based on subcarrier location and, for the secondary sidebands, the value of ASF.
4. For 16-QAM subcarriers, additionally scale the I and Q components of the complex constellation values by the 16-QAM scale factor 0.894427191. For 64-QAM subcarriers, additionally scale the I and Q components of the complex constellation value by the 64-QAM scale factor 0.43643578. These scale factors ensure that the average energies of the QPSK, 16-QAM, and 64-QAM constellations are identical.
5. Map the scaled constellation values from Steps 3 and 4 to the appropriate element of $\underline{X}$ using Table 12-19 through Table 12-23 and Figure 12-2.

Table 12-15: Signal Constellation Mapping for QPSK Data Subcarriers

I Bit	Q Bit	Constellation Value
0	0	$(-1 - j1)$
0	1	$(-1 + j1)$
1	0	$(1 - j1)$
1	1	$(1 + j1)$

Table 12-16: Signal Constellation Mapping for 16-QAM Data Subcarriers

MSB I Bit	LSB I Bit	MSB Q Bit	LSB Q Bit	Constellation Value
0	0	0	0	$(-1.5 - j1.5)$
0	0	0	1	$(-1.5 - j0.5)$
0	0	1	0	$(-1.5 + j1.5)$
0	0	1	1	$(-1.5 + j0.5)$
0	1	0	0	$(-0.5 - j1.5)$
0	1	0	1	$(-0.5 - j0.5)$
0	1	1	0	$(-0.5 + j1.5)$
0	1	1	1	$(-0.5 + j0.5)$
1	0	0	0	$(1.5 - j1.5)$
1	0	0	1	$(1.5 - j0.5)$
1	0	1	0	$(1.5 + j1.5)$
1	0	1	1	$(1.5 + j0.5)$
1	1	0	0	$(0.5 - j1.5)$
1	1	0	1	$(0.5 - j0.5)$
1	1	1	0	$(0.5 + j1.5)$
1	1	1	1	$(0.5 + j0.5)$

Table 12-17: Signal Constellation Mapping for 64-QAM Data Subcarriers

MSB I Bit	XSB I Bit	LSB I Bit	MSB Q Bit	XSB Q Bit	LSB Q Bit	Constellation Value
0	0	0	0	0	0	$(-3.5 - j3.5)$
0	0	0	0	0	1	$(-3.5 - j2.5)$
0	0	0	0	1	0	$(-3.5 - j0.5)$
0	0	0	0	1	1	$(-3.5 - j1.5)$
0	0	0	1	0	0	$(-3.5 + j3.5)$
0	0	0	1	0	1	$(-3.5 + j2.5)$
0	0	0	1	1	0	$(-3.5 + j0.5)$
0	0	0	1	1	1	$(-3.5 + j1.5)$
0	0	1	0	0	0	$(-2.5 - j3.5)$
0	0	1	0	0	1	$(-2.5 - j2.5)$
0	0	1	0	1	0	$(-2.5 - j0.5)$
0	0	1	0	1	1	$(-2.5 - j1.5)$
0	0	1	1	0	0	$(-2.5 + j3.5)$
0	0	1	1	0	1	$(-2.5 + j2.5)$
0	0	1	1	1	0	$(-2.5 + j0.5)$
0	0	1	1	1	1	$(-2.5 + j1.5)$
0	1	0	0	0	0	$(-0.5 - j3.5)$
0	1	0	0	0	1	$(-0.5 - j2.5)$
0	1	0	0	1	0	$(-0.5 - j0.5)$
0	1	0	0	1	1	$(-0.5 - j1.5)$
0	1	0	1	0	0	$(-0.5 + j3.5)$
0	1	0	1	0	1	$(-0.5 + j2.5)$
0	1	0	1	1	0	$(-0.5 + j0.5)$
0	1	0	1	1	1	$(-0.5 + j1.5)$
0	1	1	0	0	0	$(-1.5 - j3.5)$
0	1	1	0	0	1	$(-1.5 - j2.5)$
0	1	1	0	1	0	$(-1.5 - j0.5)$
0	1	1	0	1	1	$(-1.5 - j1.5)$
0	1	1	1	0	0	$(-1.5 + j3.5)$
0	1	1	1	0	1	$(-1.5 + j2.5)$
0	1	1	1	1	0	$(-1.5 + j0.5)$
0	1	1	1	1	1	$(-1.5 + j1.5)$
1	0	0	0	0	0	$(3.5 - j3.5)$
1	0	0	0	0	1	$(3.5 - j2.5)$
1	0	0	0	1	0	$(3.5 - j0.5)$
1	0	0	0	1	1	$(3.5 - j1.5)$
1	0	0	1	0	0	$(3.5 + j3.5)$

MSB I Bit	XSB I Bit	LSB I Bit	MSB Q Bit	XSB Q Bit	LSB Q Bit	Constellation Value
1	0	0	1	0	1	$(3.5 + j2.5)$
1	0	0	1	1	0	$(3.5 + j0.5)$
1	0	0	1	1	1	$(3.5 + j1.5)$
1	0	1	0	0	0	$(2.5 - j3.5)$
1	0	1	0	0	1	$(2.5 - j2.5)$
1	0	1	0	1	0	$(2.5 - j0.5)$
1	0	1	0	1	1	$(2.5 - j1.5)$
1	0	1	1	0	0	$(2.5 + j3.5)$
1	0	1	1	0	1	$(2.5 + j2.5)$
1	0	1	1	1	0	$(2.5 + j0.5)$
1	0	1	1	1	1	$(2.5 + j1.5)$
1	1	0	0	0	0	$(0.5 - j3.5)$
1	1	0	0	0	1	$(0.5 - j2.5)$
1	1	0	0	1	0	$(0.5 - j0.5)$
1	1	0	0	1	1	$(0.5 - j1.5)$
1	1	0	1	0	0	$(0.5 + j3.5)$
1	1	0	1	0	1	$(0.5 + j2.5)$
1	1	0	1	1	0	$(0.5 + j0.5)$
1	1	0	1	1	1	$(0.5 + j1.5)$
1	1	1	0	0	0	$(1.5 - j3.5)$
1	1	1	0	0	1	$(1.5 - j2.5)$
1	1	1	0	1	0	$(1.5 - j0.5)$
1	1	1	0	1	1	$(1.5 - j1.5)$
1	1	1	1	0	0	$(1.5 + j3.5)$
1	1	1	1	0	1	$(1.5 + j2.5)$
1	1	1	1	1	0	$(1.5 + j0.5)$
1	1	1	1	1	1	$(1.5 + j1.5)$

12.2.2 Reference Subcarrier Mapping Procedures

R is read one row at a time and a row of R is processed every T_s. Each row of R is a vector of bits of length 61, indexed from 0 to 60. Selected bits of this vector are mapped to reference subcarriers according to the service mode as shown in Table 12-24 and Table 12-25.

Since the output vector X contains complex values, the following steps are taken to map a row of R to an element of X:

1. Read a bit value from a row vector of R.
2. Map the bit to a complex, *binary phase shift keying* (BPSK)-modulated constellation value using Table 12-18.
3. Scale the I and Q components of the complex constellation value using the appropriate amplitude scale factor from Table 5-1 through Table 5-3 and, for secondary subcarriers, according to the state of ASF.
4. Map the scaled constellation value to the appropriate element of X using Figure 12-2, Table 12-24, and Table 12-25 for the current service mode.

Table 12-18: Signal Constellation Mapping for Reference Subcarriers

Bit Value	Constellation Value
0	$(-1 - j1)$
1	$(1 + j1)$

12.3 OFDM Subcarrier Mapper

Subsection 12.3.1 presents the mapping tables for data subcarriers. Subsection 12.3.2 presents the mapping tables for reference subcarriers.

12.3.1 Data Subcarrier Mapping Tables by Service Mode

The tables defining the data subcarrier mapping by service mode are presented in Subsection 12.3.1.1 through Subsection 12.3.1.5. The subcarrier numbers can be translated to indices of $\underline{X}$ by adding 546. For example, data subcarrier -545 maps to index 1 of $\underline{X}$.

12.3.1.1 Service Mode MP1

In service mode MP1, frequency partitions are mapped to OFDM subcarriers as presented in Table 12-19.

Table 12-19: Data Subcarrier Mapping – Service Mode MP1

Frequency Partition	Starting Subcarrier Number	Ending Subcarrier Number
0	-545	-528
1	-526	-509
2	-507	-490
3	-488	-471
4	-469	-452
5	-450	-433
6	-431	-414
7	-412	-395
8	-393	-376
9	-374	-357
18	357	374
19	376	393
20	395	412
21	414	431
22	433	450
23	452	469
24	471	488
25	490	507
26	509	526
27	528	545

12.3.1.2 Service Mode MP2

In service mode MP2, frequency partitions are mapped to OFDM subcarriers as presented in Table 12-20.

Table 12-20: Data Subcarrier Mapping – Service Mode MP2

Frequency Partition	Starting Subcarrier Number	Ending Subcarrier Number
0	-545	-528
1	-526	-509
2	-507	-490
3	-488	-471
4	-469	-452
5	-450	-433
6	-431	-414
7	-412	-395
8	-393	-376
9	-374	-357
10	-355	-338
17	338	355
18	357	374
19	376	393
20	395	412
21	414	431
22	433	450
23	452	469
24	471	488
25	490	507
26	509	526
27	528	545

12.3.1.3 Service Mode MP3

In service mode MP3, frequency partitions are mapped to OFDM subcarriers as presented in Table 12-21.

Table 12-21: Data Subcarrier Mapping – Service Mode MP3

Frequency Partition	Starting Subcarrier Number	Ending Subcarrier Number
0	-545	-528
1	-526	-509
2	-507	-490
3	-488	-471
4	-469	-452
5	-450	-433
6	-431	-414
7	-412	-395
8	-393	-376
9	-374	-357
10	-355	-338
11	-336	-319
16	319	336
17	338	355
18	357	374
19	376	393
20	395	412
21	414	431
22	433	450
23	452	469
24	471	488
25	490	507
26	509	526
27	528	545

12.3.1.4 Service Modes MP11, MP5, MP6, MP1X, DSB1, MP1XOV, MP6OV, and DSB1OV

In service modes MP11, MP5, MP6, MP1X, DSB1, MP1XOV, MP6OV, and DSB1OV, frequency partitions are mapped to OFDM subcarriers as presented in Table 12-22.

Table 12-22: Data Subcarrier Mapping – Service Modes MP11, MP5, MP6, MP1X, DSB1, MP1XOV, MP6OV, and DSB1OV

Frequency Partition	Starting Subcarrier Number	Ending Subcarrier Number
0	-545	-528
1	-526	-509
2	-507	-490
3	-488	-471
4	-469	-452
5	-450	-433
6	-431	-414
7	-412	-395
8	-393	-376
9	-374	-357
10	-355	-338
11	-336	-319
12	-317	-300
13	-298	-281
14	281	298
15	300	317
16	319	336
17	338	355
18	357	374
19	376	393
20	395	412
21	414	431
22	433	450
23	452	469
24	471	488
25	490	507
26	509	526
27	528	545

12.3.1.5 Service Mode MS5

In service mode MS5, frequency partitions are mapped to OFDM subcarriers as presented in Table 12-23.

Table 12-23: Data Subcarrier Mapping – Service Mode MS5

Frequency Partition	Starting Subcarrier Number	Ending Subcarrier Number
28	-266	-249
29	-247	-230
30	-228	-211
31	-209	-192
32	-190	-173
33	-171	-154
34	-152	-135
35	-133	-116
36	-114	-97
37	-95	-78
38	-76	-59
39	-57	-40
40	-38	-21
41	-19	-2
42	2	19
43	21	38
44	40	57
45	59	76
46	78	95
47	97	114
48	116	133
49	135	152
50	154	171
51	173	190
52	192	209
53	211	228
54	230	247
55	249	266

12.3.2 Reference Subcarrier Mapping Tables by Service Mode

The tables defining the reference subcarrier mapping by service mode are presented in Subsection 12.3.2.1 and Subsection 12.3.2.2. The reference subcarrier numbers can be translated to indices of X by adding 546. For example, reference subcarrier -546 maps to index 0 of X.

12.3.2.1 Primary Service Modes

Table 12-24 presents the mapping of columns of R to subcarriers for each primary service mode.

Table 12-24: Primary Reference Subcarrier Mapping

Subcarrier Number	Service Mode				
	MP1	MP2	MP3	MP11, MP5, MP6, MP1X, DSB1, MP1XOV, MP6OV, and DSB1OV	
-546	0	0	0	0	
-527	1	1	1	1	
-508	2	2	2	2	
-489	3	3	3	3	
-470	4	4	4	4	
-451	5	5	5	5	
-432	6	6	6	6	
-413	7	7	7	7	
-394	8	8	8	8	
-375	9	9	9	9	
-356	10	10	10	10	
-337	N/A	11	11	11	
-318	N/A	N/A	12	12	
-299	N/A	N/A	N/A	13	
-280	N/A	N/A	N/A	14	
280	N/A	N/A	N/A	46	
299	N/A	N/A	N/A	47	
318	N/A	N/A	48	48	
337	N/A	49	49	49	
356	50	50	50	50	
375	51	51	51	51	
394	52	52	52	52	
413	53	53	53	53	
432	54	54	54	54	
451	55	55	55	55	
470	56	56	56	56	
489	57	57	57	57	
508	58	58	58	58	
527	59	59	59	59	
546	60	60	60	60	

12.3.2.2 Secondary Service Modes

Table 12-25 presents the mapping of columns of $\underline{\underline{R}}$ to subcarriers for secondary service mode MS5.

Table 12-25: Secondary Reference Subcarrier Mapping

Subcarrier Number	Service Mode
	MS5
-267	15
-248	16
-229	17
-210	18
-191	19
-172	20
-153	21
-134	22
-115	23
-96	24
-77	25
-58	26
-39	27
-20	28
-1	29
0	N/A
1	31
20	32
39	33
58	34
77	35
96	36
115	37
134	38
153	39
172	40
191	41
210	42
229	43
248	44
267	45

13 OFDM Signal Generation

13.1 Introduction

OFDM Signal Generation receives complex, frequency-domain, OFDM symbols from OFDM Subcarrier Mapping, and outputs time-domain pulses representing the digital portion of the FM HD Radio signal. A conceptual block diagram of OFDM Signal Generation is shown in Figure 13-1.

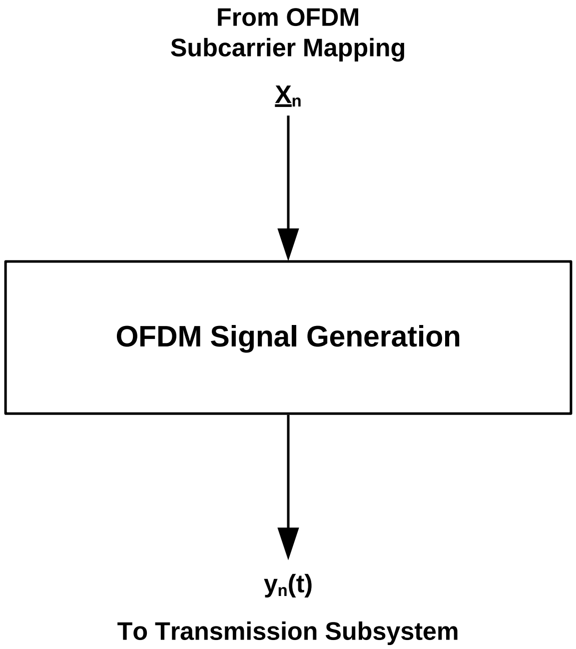


Figure 13-1: OFDM Signal Generation Conceptual Block Diagram

The input to OFDM Signal Generation is a complex vector $\underline{X}_n$ of length L, representing the complex constellation values for each OFDM subcarrier in OFDM symbol n. For notational convenience, the output of OFDM Subcarrier Mapping described in Section 12 did not use the subscript n. Rather, it referred to the vector $\underline{X}$ as representing a single OFDM symbol. In this section, the subscript is appended to $\underline{X}$ because of the significance of n to OFDM Signal Generation.

The output of OFDM Signal Generation is a complex, baseband, time-domain pulse $y_n(t)$, representing the digital portion of the FM HD Radio signal for OFDM symbol n.

13.2 Functionality

Let $\underline{X}_n[k]$ be the scaled constellation points from OFDM Subcarrier Mapping for the n^{th} symbol, where k indexes the OFDM subcarriers such that k = 0,1,2,3,…,L-1.

Let $y_n(t)$ denote the time-domain output of OFDM Signal Generation for the n^{th} symbol. Then, $y_n(t)$ is written in terms of $\underline{X}_n[k]$ as follows:

$$y_n(t) = h(t - nT_s) \cdot \sum_{k=0}^{L-1} \underline{X}_n[k] \cdot e^{j2\pi \cdot \Delta f \left[k - \frac{(L-1)}{2}\right] \cdot (t - nT_s)}$$

where n = 0,1,2,3,…,∞ and $0 \le t < \infty$.

L = 1093 is the total number of OFDM subcarriers.

T_s and Δf are the OFDM symbol duration and OFDM subcarrier spacing, respectively, as defined in Subsection 3.5.

The *pulse-shaping function* $h(\xi)$ is defined as:

$$h(\xi) = \begin{cases} \cos\left(\pi \dfrac{\alpha T - \xi}{2\alpha T}\right) & if \quad 0 < \xi < \alpha T \\ 1 & if \quad \alpha T \le \xi \le T \\ \cos\left(\pi \dfrac{T - \xi}{2\alpha T}\right) & if \quad T < \xi < T(1+\alpha) \\ 0 & elsewhere \end{cases}$$

where α is the cyclic prefix width defined in Subsection 3.5.

$T = 1/\Delta f$ is the reciprocal of the OFDM subcarrier spacing.

14 Transmission Subsystem

14.1 Introduction

The Transmission Subsystem formats the baseband FM HD Radio waveform for transmission through the VHF channel. Functions include symbol concatenation and frequency up-conversion. In addition, when transmitting the Hybrid or Extended Hybrid waveforms, this function modulates the baseband analog signal before combining it with the digital waveform.

The input to this module is a complex, baseband, time-domain OFDM symbol, $y_n(t)$, from the OFDM Signal Generation function. When transmitting the Hybrid or Extended Hybrid waveform, a diversity-delayed (T_{ad}), baseband, analog signal m(t), plus an appropriate implementation-specific *Transmit Audio Alignment Delay*, T_{T5a}, plus optional Subsidiary Communications Authorization (SCA) signals, are also inputs to this module. Note that diversity delay is applied to the analog signal for only those hybrid and extended hybrid service modes that support blending. The output of this module is the VHF FM HD Radio waveform.

Refer to Figure 14-1 for a functional block diagram of the All Digital Transmission Subsystem; refer to Figure 14-2 for a functional block diagram of the Hybrid and Extended Hybrid Transmission Subsystems.

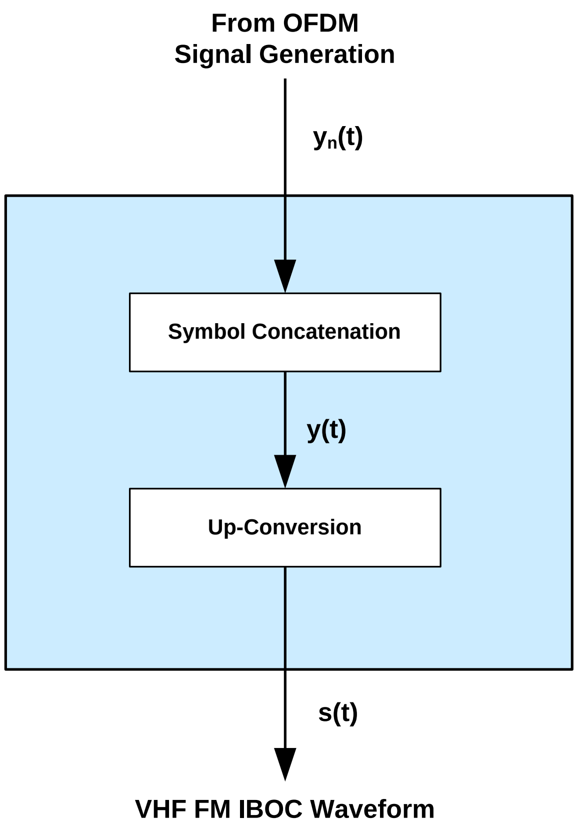


Figure 14-1: All Digital Transmission Subsystem Functional Block Diagram

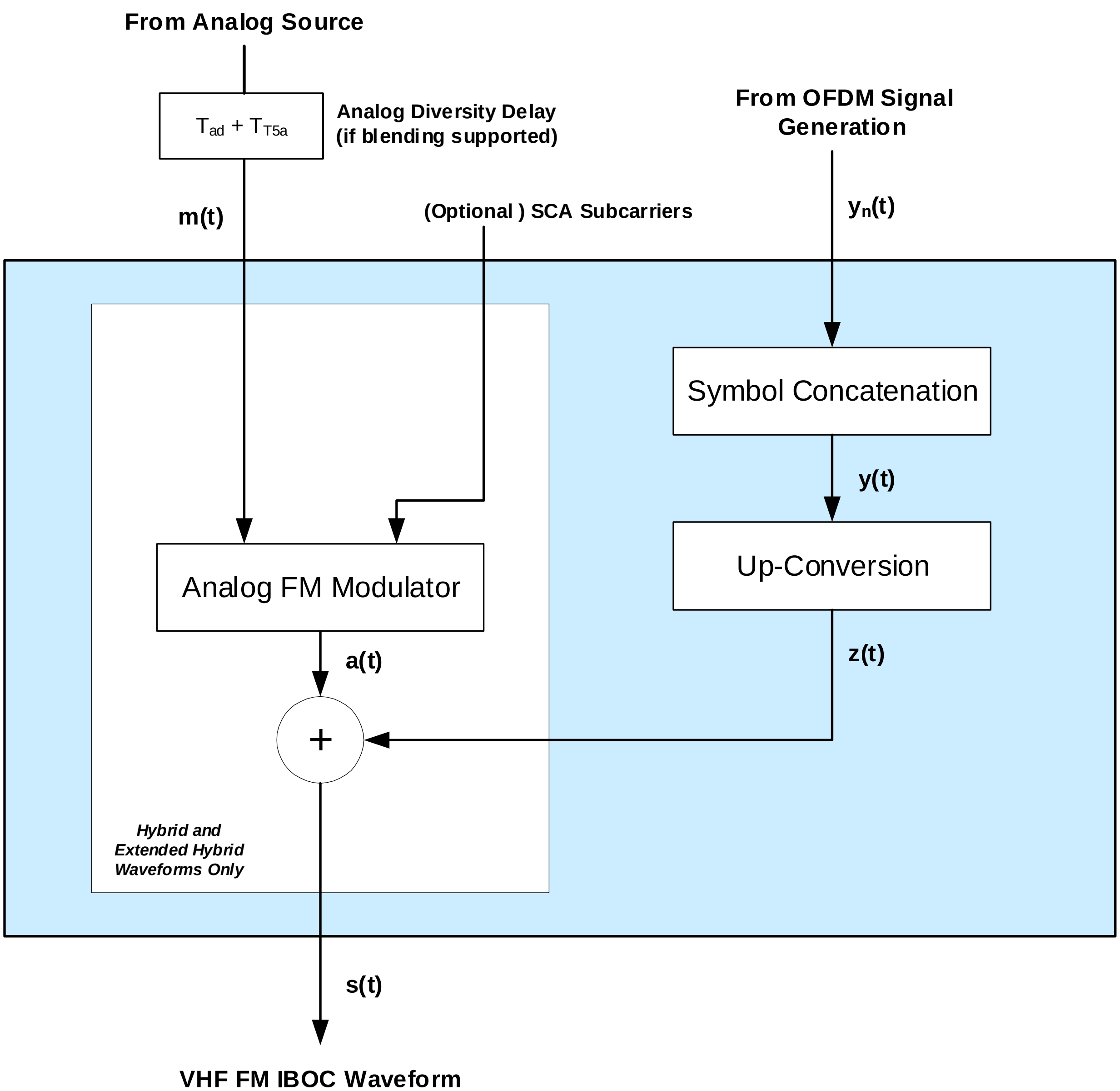


Figure 14-2: Hybrid/Extended Hybrid Transmission Subsystem Functional Block Diagram

14.2 Functional Components

The functional components of the Transmission Subsystem are specified in Subsection 14.2.1 through Subsection 14.2.5.

14.2.1 Symbol Concatenation

The individual time-domain OFDM symbols generated by OFDM Signal Generation are concatenated to produce a continuum of pulses over t = 0,1,2,3,…,∞ as follows:

$$y(t) = \sum_{n=0}^{\infty} y_n(t)$$

14.2.2 Up-Conversion

The concatenated digital signal y(t) is translated from baseband to the RF carrier frequency as follows:

$$z(t) = \mathrm{Re}\left(e^{-j2\pi f_c t} \cdot y(t) \right)$$

where j is the unitary imaginary number, f_c is the VHF *allocated channel* frequency, and Re( ) denotes the real component of the complex quantity. For the All Digital waveform, the output of the up-converter is the transmitted VHF FM HD Radio waveform, and therefore, s(t) = z(t).

The negative exponent in the up-conversion equation indicates that the RF spectrum is inverted in comparison to the baseband spectrum. This means that the negative-numbered subcarriers / lower sideband occupy the higher frequencies within the RF channel. Similarly, the positive-numbered subcarriers / upper sideband occupy the lower frequencies within the RF channel.

The carrier frequency spacing and channel numbering scheme are compatible with Title 47 CFR §73.201 (see Reference [12]). The carriers retain their 200-kHz spacing over the 88.0- to 108.0-MHz frequency range. Channels are numbered from 201 to 300 where channel 201 is centered on 88.1 MHz and channel 300 is centered on 107.9 MHz. The absolute accuracy of the carrier frequency is defined in Reference [7].

14.2.3 Analog Diversity Delay

When broadcasting the Hybrid and Extended Hybrid waveforms, the digital signal is combined with the analog FM signal as shown in Figure 14-2. However, analog diversity delay is first applied to the baseband analog FM signal for those service modes that supporting blending.

In the HD Radio system, the analog and digital signals carry the same audio program with the analog audio delayed from the corresponding digital audio at the output of the analog/digital combiner. This delay consists of a fixed portion T_{ad}, as defined in Section 3.5 (FM System Parameters), plus an adjustable portion T_{T5a}. The delay is adjusted so that the audio content in the analog and digital paths has a time diversity of precisely T_{ad} at the transmit antenna. This delay accounts for processing delay differences in the two signal paths.

The absolute accuracy of the analog diversity delay, when enabled, is defined in [7].

Ball-game mode: A radio station can disable the analog diversity delay for specialized broadcasts. The state of the analog diversity delay is indicated by the Blend Control bits in the Audio Transport layer (See Reference [4]). However, changing the state of the analog diversity delay may result in a discontinuity during reception as the receiver blends from analog to digital. Some receivers may disable digital reception entirely when analog diversity delay is disabled.

14.2.4 Analog FM Modulator

For the Hybrid and Extended Hybrid waveforms, the baseband analog signal m(t) is frequency modulated to produce an RF analog FM waveform identical to existing analog signals. The FM-modulated analog signal, including any SCAs, will maintain compatibility with Title 47 CFR Part 73, Subparts B, C, and H. In addition, the analog signal will be compatible with the Emergency Alert System (EAS) as specified in Title 47 CFR Part 11 (See Reference [11]).

14.2.5 Analog/Digital Combiner

When broadcasting the Hybrid or Extended Hybrid waveform, the analog-modulated FM RF signal is combined with the digitally modulated RF signal to produce the VHF FM HD Radio waveform, s(t). Both the analog and digital portions of the waveform are centered on the same carrier frequency.

The levels of each digital sideband in the output spectrum are appropriately scaled by OFDM Subcarrier Mapping. These scale factors, as well as the ratio of the total power in the analog FM signal to the total power in the digital sidebands, are provided in Reference [7].

The spectral emissions limits of the composite HD Radio RF signal are defined in Reference [7].

15 Glossary

In order to better understand the terms and concepts in this document, the following definitions apply:

Absolute L1 Frame Number (ALFN)
A number assigned to each transmitted L1 frame that provides a reference to absolute time. The start of ALFN 0 occurred at 00:00:00 Coordinated Universal Time (UTC) on January 6, 1980. The start of every subsequent L1 frame occurs at an exact integer multiple of T_f after that instant in time.

All Digital waveform
The transmitted waveform composed entirely of digitally modulated subcarriers (subcarriers -546 to +546) without an analog FM signal. Use of this waveform will normally follow an initial transitional phase utilizing Hybrid waveforms incorporating both analog and digital modulation. (See *Hybrid waveform* and *Extended Hybrid waveform.*)

allocated channel
One of the one hundred possible frequency assignments in the FM band, as defined in Reference [12].

amplitude modulation (AM)
Modulation in which the amplitude of a carrier wave is varied in accordance with the amplitude of the modulating signal.

amplitude scale factor
A factor which multiplies the baseband components of a particular OFDM subcarrier of the transmitted spectrum to constrain the radiated power to a prescribed level.

analog signal
Refers to signals that are modulated on the main carrier by conventional high-modulation-index frequency modulation. (See *digital signal.*)

backup
Low-latency, delayed component of a time-diverse logical channel.

BC range
The range of L1 Blocks, m1:m2, spanned by a transfer frame, indicating its position within an L1 frame.

binary phase shift keying (BPSK)
A form of digital phase modulation that assigns one of two discrete phases, differing by 180 degrees, to the carrier. Each BPSK symbol conveys one bit of information.

channel encoding
The process used to add redundancy to each of the logical channels to improve the reliability of the transmitted information.

characterization parameters
The unique set of defining parameters for each logical channel for a given service mode. The modulation, channel encoding, interleaving, spectral mapping, and diversity delay of the logical channel determine its characterization parameters.

code rate
Defines the increase in overhead on a coded channel resulting from channel encoding. It is the ratio of information bits to the total number of bits after coding.

column pair
Bits from adjacent columns in an interleaver partition that represent the I and Q bit pair to map to a symbol.

convolutional encoding
A form of forward-error-correction channel encoding that inserts coding bits into a continuous stream of information bits to form a predictable structure. Unlike a block encoder, a convolutional encoder has memory; its output is a function of current and previous inputs.

Configuration Administrator
The Configuration Administrator is a system function that configures each of the layers using SCCH information or parameters which do not change often.

differential encoding
Encoding process in which signal states are represented as changes to succeeding values rather than absolute values.

digital signal
Refers to signals that are digitally modulated on subcarriers by OFDM (q.v.). (See *analog signal*.)

diversity delay
Imposition of a fixed time delay in one of two channels carrying the same information to defeat non-stationary channel impairments such as fading and impulsive noise.

Extended Hybrid waveform
The transmitted waveform composed of the analog FM signal plus digitally modulated primary main subcarriers (subcarriers +356 to +546 and -356 to -546) and some or all primary extended subcarriers (subcarriers +280 to +355 and -280 to -355). This waveform could be used during an initial transitional phase preceding conversion to the All Digital waveform. (See *All Digital waveform* and *Hybrid waveform.*)

fading
The variation (with time) of the amplitude or relative phase (or both) of one or more frequency components of a received signal.

frequency modulation (FM)
Modulation in which the instantaneous frequency of a sine wave carrier is caused to depart from the channel center frequency by an amount proportional to the instantaneous amplitude of the modulating signal.

frequency partition
A group of 19 OFDM subcarriers containing 18 data subcarriers and one reference subcarrier.

Frequency Partition Mapper

Function of OFDM Subcarrier Mapping that assigns interleaver partitions to frequency partitions. Depending on modulation order, one (QPSK), two (16-QAM), or three (64-QAM) interleaver partitions are mapped to each frequency partition.

hierarchical modulation

See “layered modulation”

Hybrid waveform
The transmitted waveform composed of the analog FM-modulated signal, plus digitally modulated Primary Main subcarriers (subcarriers +356 to +546 and -356 to -546). This waveform could be used during an initial transitional phase preceding conversion to the All Digital waveform. (See *All Digital waveform* and *Extended Hybrid waveform*.)

interlace
The process of alternately extracting bits, row by row, from a selected block of main and backup interleaver matrices and writing them into a single composite interleaver matrix.

interleaver block
A logical subdivision of an interleaver partition. Each interleaver block contains 32 rows and 36 columns.

interleaver depth
The number of rows in an interleaver matrix. The system employs four interleaver depths: L1 block (32 rows); L1 block pair (64 rows); L1 frame (512 rows); L1 frame pair (1024 rows).

interleaver matrix
A two-dimensional array containing the output of an interleaving process.

interleaver partition
A logical subdivision of the overall interleaver matrix. Each interleaver partition contains 36 columns and 32·B rows where B is the number of interleaver blocks.

interleaving
A reordering of the message bits to distribute them in time (over different OFDM symbols) and frequency (over different OFDM subcarriers) to mitigate the effects of signal fading and interference.

interleaving process
A series of manipulations performed on one or more coded transfer frames (vectors) to reorder their bits into one or more interleaver matrices whose contents are destined for a particular portion of the transmitted spectrum.

L1 block
A unit of time of duration T_b. Each L1 frame is comprised of 16 L1 blocks.

L1 Block Count
An index that indicates one of 16 equal subdivisions of an L1 frame.

L1 block pair
Two contiguous L1 blocks. A unit of time duration T_p.

L1 block pair rate

The rate, equal to the reciprocal of the L1 block pair duration, $\left(\frac{1}{T_p}\right)$, at which selected transfer frames are conducted through Layer 1.

L1 block rate

The rate, equal to the reciprocal of the L1 block duration, $\left(\frac{1}{T_b}\right)$, at which selected transfer frames are conducted through Layer 1.

L1 frame
A specific time slot of duration T_f identified by an ALFN. The transmitted signal may be considered to consist of a series of L1 frames.

L1 frame pair
Two contiguous L1 frames. A unit of time duration $2 \cdot T_f$

L1 frame rate
The rate, equal to the reciprocal of the L1 frame duration $\left(\frac{1}{T_f}\right)$, at which selected transfer frames are conducted through Layer 1.

latency
The time delay that a logical channel imposes on a transfer frame as it traverses Layer 1. One of the three characterization parameters. (See *robustness* and *transfer*.)

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 FM radio link at the specified service mode.

Layer 2 (L2)
The Channel Mux 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.

Layer 2 protocol data units (L2 PDUs)
Units of user content and upper layer protocol control information transferred from Layer 2 to Layer 1.

layered modulation
A backward-compatible means of significantly enhancing capacity by simultaneously broadcasting two service modes, one of which is “overlayed” on top of an existing “base” service mode. Legacy receivers detect the base layer, while newer advanced receivers also detect the overlay for additional capacity.

logical channel
A signal path that conducts transfer frames from Layer 2 through Layer 1 with a specified grade of service.

lower sideband
The group of OFDM subcarriers (subcarriers number -1 through -546) below the carrier frequency.

main
High-throughput, deeply interleaved, non-delayed component of a time-diverse logical channel.

modulation
The process of varying the characteristics (e.g., amplitude, frequency, phase, etc.) of a periodic carrier signal using a modulating signal containing information to be transmitted.

modulation component
An indicator of the relative position of constituent I and Q bit pairs within the indices to the complex constellation values that are assigned to OFDM data subcarriers in OFDM Symbol Mapping.

mother code
The complete code sequence generated by a convolutional encoder. (See *puncturing*.)

mother codeword
A code sequence generated by a convolutional encoder. (See *puncturing*.)

OFDM Signal Generation
The function that generates the modulated baseband signal in the time domain.

OFDM subcarrier
A discrete frequency-domain signal within the allocated channel that encodes digital data through its amplitude and/or phase. The total set of subcarriers, taken in aggregate for a period of T_s, provides the digital data for that time interval. (See *OFDM symbol*.)

OFDM Subcarrier Mapping
The function that assigns the interleaved logical channels (interleaver partitions) to the OFDM subcarriers (frequency partitions).

OFDM symbol
Time domain pulse of duration T_s, representing all the active subcarriers and containing all the data in one row from the interleaver and system control data sequence matrices. The transmitted waveform is the concatenation of successive OFDM symbols.

Orthogonal Frequency Division Multiplexing (OFDM)
A parallel multiplexing scheme that modulates a data stream onto a large number of orthogonal subcarriers that are transmitted simultaneously. (See *OFDM symbol.*)

parity
In binary-coded data, a condition maintained so that in any permissible coded expression, the total number of “1”s or “0”s is always odd, or always even.

Primary Extended (PX) sideband
The portion of the primary sideband that holds the additional frequency partitions (1, 2, or 4) inside the main partitions in the FM Extended Hybrid and All Digital waveforms. It consists, at most, of subcarriers 280 through 355 and -280 through -355.

Primary Main (PM) sidebands
The ten partitions in the primary sideband consisting of subcarriers 356 through 545 and -356 through -545.

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.

pulse-shaping function
A time-domain pulse superimposed on the OFDM symbol to improve its spectral characteristics.

puncturing
The process of removing selected bits from the mother codeword to increase FEC code rate.

Quadrature Phase Shift Keying (QPSK)
A form of digital phase modulation that assigns one of four discrete phases, differing by 90 degrees, to the carrier. Each QPSK symbol conveys two bits of information.

Quadrature Amplitude Modulation (QAM)
A form of digital modulation that assigns one of several discrete phases and amplitudes to the carrier. 4-QAM (QPSK) symbols convey two bits of information over four constellation points, 16-QAM symbols convey four bits of information across sixteen constellation points, and 64-QAM symbols convey six bits of information across sixty-four constellation points.

reference subcarrier
A dedicated OFDM subcarrier in L1 of the HD Radio system used to convey L1 system control and status information contained in the system control data sequence. The number of reference subcarriers broadcast in a given waveform depends on the service mode. Reference subcarriers are typically used by receivers as an acquisition and synchronization aid.

robustness
The ability of a logical channel to withstand channel impairments such as noise, interference, and fading. There are nine distinct levels of robustness designed into Layer 1 of the FM air interface. One of the three characterization parameters. (See *latency* and *transfer*.)

scrambling
The process of modulo 2 summing the input data bits with a pseudo-random bit stream to randomize the time domain bit stream.

secondary sidebands
The sidebands to be added in the spectrum vacated by the analog signal. The secondary sidebands are divided into upper and lower sidebands, each containing 14 frequency partitions. The secondary sidebands consist of subcarriers −267 through +267.

service mode
A specific configuration of operating parameters specifying throughput, performance level, and selected logical channels.

Signal Constellation Mapper
The function in OFDM Subcarrier Mapping that associates I, Q bit pairs with specific states, or associates single bits from $\underline{\underline{R}}$ with specific BPSK states.

spectral emissions mask
A specification setting the maximum level of out-of-band components of the transmitted signal.

spectral mapping
The association of specific logical channels with specific subcarriers or groups of subcarriers.

submatrix
A matrix extracted from a larger matrix; one or more of its dimensions is less than that of the larger matrix

system control
Data from the Configuration Administrator conveying control such as service mode, primary amplitude scale factors, and secondary amplitude scale factor select.

System Control Channel (SCCH)
A channel which transports control information from the Configuration Administrator to Layer 1 and also conveys status information from Layer 1 to Layer 2, through the system control processing.

system control data sequence
A sequence of bits destined for each reference subcarrier representing the various system control components relayed between the Configuration Administrator and Layer 1.

system control processing
The function that generates the system control data sequence.

system protocol stack
The protocols associated with operation of the layers of the HD Radio system.

system time alignment, T_{st}
Internal time delay to absorb variations in internal processing time to maintain message alignment with L1 blocks Frames.

transfer
A measure of the data throughput through a logical channel. One of the three characterization parameters. (See *latency* and *robustness*.)

transfer frame
An ordered, one-dimensional collection of data bits of specified length grouped for processing through a logical channel for exchange with the physical layer.

transfer frame modulus
The number of transfer frames in an L1 frame.

transfer frame multiplexer
A device that combines two or more transfer frames into a single vector.

transfer frame number
A number, $F_{m1:m2}^{n}$, that specifies the ALFN, n, and BC range, m1:m2, associated with a particular transfer frame, in order to relate the transfer frame to absolute time.

transfer frame rate
The number of transfer frames per second

transfer frame size
The number of bits in a transfer frame.

transmission subsystem
The functional component used to format and up-convert the baseband HD Radio waveform for transmission through the very-high frequency (VHF) channel.

transmit time alignment, T_{T1a}
An adjustment applied to make the digital time diversity between P1 and P1′ and S1 and S1′ be precisely T_{dd} at the transmit antenna.

transmit audio alignment, T_{T5a}
Adjusted so that the audio content in the analog and digital paths has a time diversity of precisely T_{ad} at the TX antenna (applies to hybrid service modes only)

upper sideband
The group of OFDM subcarriers (subcarrier numbers 1 through +546) above the carrier frequency.

vector
A one-dimensional array.

HD Radio™ Air Interface Design Description Layer 1 AM

Rev. G

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TRADEMARKS

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Table of Contents

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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 defines the generation of Layer 1 AM HD Radio signals for transmission over the air to receiving equipment. It describes how control and information are passed through the Layer 1 AM air interface to generate an HD Radio signal. It focuses on the creation of the transmitted AM HD Radio signal; specific hardware and software implementation is not described.

2 Referenced Documents

STATEMENT
Each referenced document that is mentioned in this document shall be listed in the following iBiquity document:

- Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard Document Number: SY_REF_2690s

3 Abbreviations, Symbols, and Conventions

3.1 Introduction

Section 3 presents the following items pertinent to a better understanding of this document:

- Abbreviations and Acronyms
- Presentation Conventions
- Mathematical Symbols
- AM System Parameters

Note: A glossary defining the technical terms used herein is provided at the end of this document.

3.2 Abbreviations and Acronyms

AAB	Analog Audio Bandwidth Control
AABI	Analog Audio Bandwidth Indicator
ALFN	Absolute L1 Frame Number
AM	Amplitude Modulation
BC	L1 Block Count
BPSK	Binary Phase Shift Keying
FM	Frequency Modulation
GCS	Grounded Conductive Structures
GPS	Global Positioning System
HPP	High-Power PIDS Control
HPPI	High-Power PIDS Indicator
IBOC	In-band On-channel
IDS	IBOC Data Service
kbit/s	kilobits per second
L1	Layer 1
L2	Layer 2
LC	Logical Channel
MA1	AM Hybrid Service Mode 1
MA3	AM All Digital Service Mode 3
MF	Medium Frequency
N/A	Not Applicable
OFDM	Orthogonal Frequency Division Multiplexing
P1–P3	Primary Logical Channels 1 through 3
PDU	Protocol Data Unit
PIDS	Primary IBOC Data Service Logical Channel
PL	Power Level Control
PLI	Power Level Indicator
PSM	Service Mode Control
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RDB	Reduced Digital Bandwidth Control
RDBI	Reduced Digital Bandwidth Indicator
RF	Radio Frequency
SCCH	System Control Channel

SIS	Station Information Service
SMI	Service Mode Indicator
UTC	Coordinated Universal Time
VHF	Very High Frequency

3.3 Presentation Conventions

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

- Glossary terms are presented in italics upon their first usage in the text.
- 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.

3.4 Mathematical Symbols

3.4.1 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 component 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	$\underline{\underline{u}}[i][j]$, $\underline{\underline{V}}[i][j]$
$\begin{bmatrix} \quad & \quad \\ \quad & \quad \end{bmatrix}$	Indicates the components 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]

3.4.2 Arithmetic Operators

The arithmetic operators defined below are used throughout this document.

<table>
<tr><th>Category</th><th>Definition</th><th>Examples</th></tr>
<tr><td>·</td><td>Indicates a multiplication operation</td><td>3·4 = 12</td></tr>
<tr><td>INT( )</td><td>Indicates the integer portion of a real number</td><td>INT(5/3) = 1
INT(-1.8) = -1</td></tr>
<tr><td>a MOD b</td><td>Indicates a modulo operation</td><td>33 MOD 16 = 1</td></tr>
<tr><td>$\oplus$</td><td>Indicates modulo-2 binary addition</td><td>$1 \oplus 1 = 0$</td></tr>
<tr><td>|</td><td>Indicates the concatenation of two vectors</td><td>$\underline{A}$ = [$\underline{B}$ | $\underline{C}$]
The resulting vector $\underline{A}$ consists of the elements of $\underline{B}$ followed by the elements of $\underline{C}$.</td></tr>
<tr><td>j</td><td>Indicates the square-root of -1</td><td>j = $\sqrt{-1}$</td></tr>
<tr><td>Re( )</td><td>Indicates the real component of a complex quantity</td><td>If x = (3 + j4), Re(x) = 3</td></tr>
<tr><td>Im( )</td><td>Indicates the imaginary component of a complex quantity</td><td>If x = (3 + j4), Im(x) = 4</td></tr>
<tr><td>$\log_{10}$</td><td>Indicates the base-10 logarithm</td><td>$\log_{10}(100) = 2$</td></tr>
<tr><td>*</td><td>Indicates complex conjugate</td><td>If x = (3 + j4), x* = (3 - j4)</td></tr>
<tr><td>| x |</td><td>Indicates the absolute value of x</td><td>| -5 | = 5
| 3 – 4 | = 1</td></tr>
</table>

3.5 AM System Parameters

The AM system parameters defined below are used throughout this document.

Parameter Name	Symbol	Units	Exact Value	Computed Value (To 4 significant figures)
OFDM Subcarrier Spacing	Δf	Hz	1488375/8192	181.7
Cyclic Prefix Width	α	none	7/128	5.469×10^{-2}
OFDM Symbol Duration	T_s	s	$(1+\alpha)/\Delta f = (135/128)\cdot(8192/1488375)$	5.805×10^{-3}
OFDM Symbol Rate	R_s	Hz	$= 1/T_s$	172.3
L1 Frame Duration	T_f	s	$65536/44100 = 256\cdot T_s$	1.486
L1 Frame Rate	R_f	Hz	$= 1/T_f$	6.729×10^{-1}
L1 Block Duration	T_b	s	$= 32\cdot T_s$	1.858×10^{-1}
L1 Block Rate	R_b	Hz	$= 1/T_b$	5.383
Digital Diversity Delay Frames	N_{dd}	none	3	3
Digital Diversity Delay Time	T_{dd}	s	$= N_{dd}\cdot T_f$	4.458
Analog Diversity Delay Time	T_{ad}	s	$4.5\cdot T_f$	6.687

4 Overview

4.1 Introduction

Layer 1 of the AM system converts information from *Layer 2* (L2) and system control from the *Configuration Administrator* into an AM HD Radio waveform for transmission in the existing allocation in the MF band. Information and control are transported in discrete *transfer frames* via multiple *logical channels*. These transfer frames are referred to as *Layer 2 Protocol Data Units* (PDUs).

The L2 PDUs vary in size and format depending on the *service mode*. The service mode, a major component of system control, determines the transmission characteristics of each logical channel. After assessing the requirements of their candidate applications, higher protocol layers select service modes that most suitably configure the logical channels. The plurality of logical channels reflects the inherent flexibility of the system, which supports simultaneous delivery of various classes of digital audio and data.

Layer 1 also receives system control from the Configuration Administrator for use by the Layer 1 System Control Processor.

This section presents the following:

- An overview of the waveforms and spectra
- An overview of the system control, including the available service modes
- An overview of the logical channels
- A high-level discussion of each of the functional components included in the Layer 1 AM air interface

Note: Throughout this document, various system parameters are globally represented as mathematical symbols. Refer to Subsection 3.5 for their values.

4.2 Waveforms and Spectra

The design provides a flexible means of transitioning to a digital broadcast system by providing two new waveform types: *Hybrid* and *All Digital*. The Hybrid waveform retains the analog AM signal, while the All Digital waveform does not. Both new waveform types conform to the current *spectral emissions mask*.

The digital signal is modulated using *Orthogonal Frequency Division Multiplexing* (OFDM). OFDM is a parallel modulation scheme in which the data stream modulates a large number of orthogonal subcarriers that are transmitted simultaneously. OFDM is inherently flexible, readily allowing the mapping of logical channels to different groups of subcarriers.

Refer to Section 5 for a detailed description of the spectra of the two waveform types.

4.2.1 Hybrid Waveform

In the Hybrid waveform, the digital signal is transmitted in *primary* and *secondary* sidebands on both sides of the host *analog signal*, as well as underneath the host analog signal in *tertiary sidebands*. The bandwidth of the analog audio can be either 5 kHz or 8 kHz, as shown in Figure 5-1 and Figure 5-2 respectively (see Subsection 5.3). If the 8 kHz bandwidth is selected, then the secondary sidebands are also underneath the host analog signal. A reduced digital bandwidth configuration is also provided where an analog audio bandwidth of up to 9.4 kHz can be accommodated, as shown in Figure 5-3.

The levels of the *OFDM subcarriers* within each *primary sideband* are all the same. However, the levels of the two primary sidebands are adjustable independently.

The secondary and tertiary sideband levels may be set to one of two selectable values. In addition, there are two reference subcarriers for system control whose levels are fixed at a value that is different from the sidebands.

The analog host is a monophonic signal. The Hybrid system does not support analog AM stereo transmissions.

4.2.2 All Digital Waveform

The greatest system enhancements are realized with the All Digital waveform. In this waveform, the analog signal is replaced with the primary sidebands whose power is increased relative to the Hybrid waveform levels. In addition, the secondary and tertiary sidebands are moved to both sides of the primary sidebands, and their power is also increased relative to the Hybrid levels. The end result is a higher power digital signal with an overall bandwidth reduction. These changes provide a more robust digital signal that is less susceptible to adjacent channel interference. Reference subcarriers are also provided to convey system control information. Their levels are fixed at a value that is different than that of the sidebands.

4.3 System Control Channel

The *system control channel* (SCCH) transports control and status information. The *service mode control* (PSM), *analog audio bandwidth control* (AAB), and *power level control* (PL) are input to Layer 1 from the Configuration Administrator, while status information is sent from Layer 1 to Layer 2.

Two service modes dictate all permissible configurations of the logical channels. They are:

- Hybrid service mode MA1
- All Digital service mode MA3

Refer to Section 6 for a detailed description of the SCCH and Section 11 for a detailed description of *system control processing*.

4.4 Logical Channels

A logical channel is a signal path that conducts L2 PDUs in transfer frames into and out of Layer 1 with a specific grade of service, determined by service mode. Layer 1 of the AM air interface provides three logical channels to higher layer protocols: P1, P3, and PIDS. P1 and P3 are intended for general purpose audio and data transfer, while PIDS is designed to carry the Station Information Service (SIS) data. The P1 logical channel is designed to be more robust than the P3 logical channel. This allows a transfer of information that can be tailored to conform to a number of diverse applications.

The approximate information rates of the three logical channels for each of the service modes are shown in Table 4-1. Calculation of the exact rates is explained in Section 7.

Table 4-1: Approximate Information Rate of AM Logical Channels

Service Mode	Approximate Channel Information Rate (kbit/s)			Waveform
	P1	P3	PIDS	
MA1	20	16	0.4	Hybrid
MA3	20	20	0.4	All Digital

The performance of each logical channel is completely described through three *characterization parameters*: *transfer*, *latency*, and *robustness*. *Channel encoding*, *spectral mapping*, *interleaver depth*, and digital *diversity delay* are the components of these characterization parameters. The service mode uniquely configures these components for each active logical channel, thereby allowing the assignment of appropriate characterization parameters.

In addition, the service mode specifies the framing and synchronization of the transfer frames through each active logical channel. Refer to Section 7 for a detailed description of the logical channels and their configuration.

4.5 Functional Components

This subsection includes a high-level description of each Layer 1 functional block and the associated signal flow. Figure 4-1 is a functional block diagram of the Layer 1 processing. Some processing stages shown in Figure 4-1 are denoted by a logical channel subscript. For example, logical channel designations are subscripted with an "S" after *scrambling* and with a "G" after channel encoding. The single underline notation for a logical channel name refers to the fact that data is passed between the various functions as *vectors*. Each logical channel has a dedicated scrambler and channel encoder. The configuration administrator is a system function that configures each of the layers using SCCH information or parameters which do not change often. However, dynamic SCCH parameters such as the L1 Block Count and ALFN are sent from Layer 1 to Layer 2.

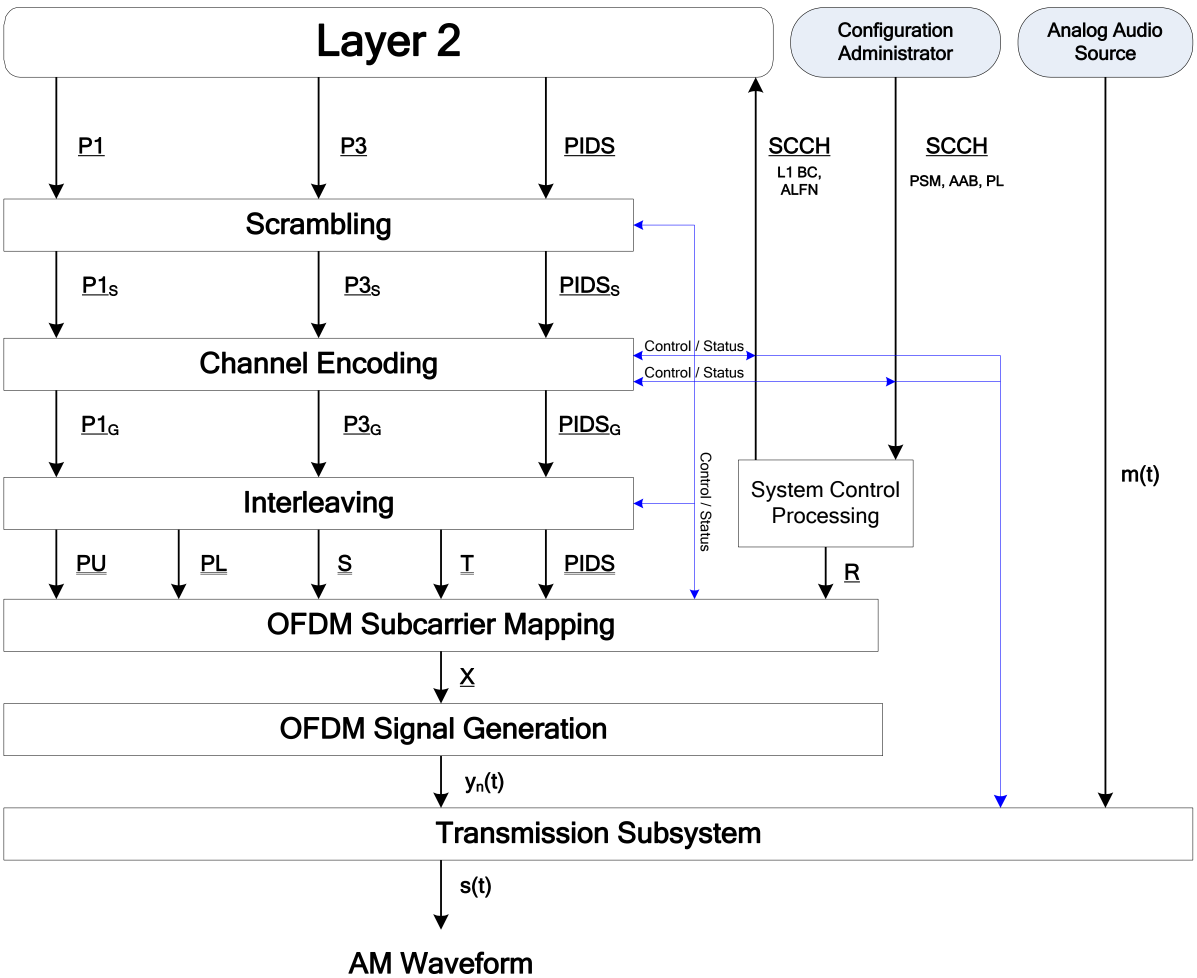

Figure 4-1: AM Air Interface L1 Functional Block Diagram

4.5.1 Scrambling

This function randomizes the digital data carried in each logical channel to mitigate signal periodicities. At the output of scrambling, the logical channel vectors retain their identity but are distinguished by the "S" subscript (for example, "$\underline{P1}_S$"). Refer to Section 8 for a detailed description of the scrambling functional component.

4.5.2 Channel Encoding

This function uses *convolutional encoding* to add redundancy to the digital data in each logical channel to improve its reliability in the presence of channel impairments. The size of the logical channel vectors is increased in inverse proportion to the *code rate*. The encoding techniques are configurable by service mode. At the output of the channel encoder, the logical channel vectors retain their identity, but are distinguished by the "G" subscript (for example, "$\underline{P1}_G$"). Refer to Section 9 for a detailed description of the channel encoding functional component.

4.5.3 Interleaving

Interleaving in time and frequency is employed to mitigate the effects of burst errors. The interleaving techniques are tailored to the MF non-uniform interference environment and are configurable by service mode. In this process, the logical channels lose their identity. The interleaver output is structured in a matrix format. Each matrix consists of information from whole or partial logical channels and is associated with a specific portion of the transmitted spectrum. The *interleaver matrix* designations reflect the spectral mapping; "$\underline{\underline{PU}}$" and "$\underline{\underline{PL}}$", for example, map to the primary sidebands while S and T map to the secondary and tertiary sidebands respectively. Digital diversity delay is also imposed on selected logical channels. Refer to Section 10 for a detailed description of the interleaving functional component.

4.5.4 System Control Processing

This function generates a vector of *system control data sequences* that includes system control information received from the Configuration Administrator (such as service mode), and status for broadcast on the reference subcarriers. This data vector is designated "$\underline{R}$" for "reference." Refer to Section 11 for a detailed description of the system control processing functional component.

4.5.5 OFDM Subcarrier Mapping

This function assigns the interleaver matrices and system control vector to OFDM subcarriers. One row of each active interleaver matrix and one bit of the system control vector is processed each *OFDM symbol* (every T_S seconds) to produce one output vector $\underline{X}$, which is a frequency-domain representation of the signal. The mapping is specifically tailored to the non-uniform interference environment encountered in the AM band and is a function of the service mode. Refer to Section 12 for a detailed description of the *OFDM subcarrier mapping* functional component.

4.5.6 OFDM Signal Generation

This function generates the digital portion of the time-domain AM HD Radio waveform. The input vectors $\underline{X}$ are transformed into a shaped time-domain baseband pulse, $y_n(t)$, defining one OFDM symbol. Refer to Section 13 for a detailed description of the *OFDM Signal Generation* functional component.

4.5.7 Transmission Subsystem

This function formats the baseband waveform for transmission through the MF channel. Major sub-functions include *symbol concatenation,* and frequency up-conversion. When transmitting the Hybrid waveform, this function modulates the AM analog audio source and combines it with the digital signal to form a composite Hybrid signal, s(t), ready for transmission. Refer to Section 14 for a detailed description of the *transmission subsystem* functional components.

5 Waveforms and Spectra

5.1 Introduction

This section describes the output spectrum for Hybrid and All Digital waveforms. Each spectrum is divided into several sidebands, which represent various subcarrier groupings. All spectra are represented at baseband.

5.2 Spectral Conventions

Each spectrum described in the following subsections shows the subcarrier number and center frequency of certain key OFDM subcarriers. The center frequency of a subcarrier is calculated by multiplying the subcarrier number by the OFDM subcarrier spacing Δf. The center of subcarrier 0 is located at 0 Hz. In this context, the center frequency is relative to the radio frequency (RF) *allocated channel*.

5.3 Hybrid Spectrum

The digital signal is transmitted in primary and secondary sidebands on both sides of the analog host signal, as well as in tertiary sidebands beneath the analog host signal as shown in Figure 5-1. In this configuration, the analog audio bandwidth is limited to 5 kHz. If the bandwidth of the analog audio exceeds 5 kHz, the secondary sidebands are also beneath the analog host signal as shown in Figure 5-2 (See Subsection 6.4). In this configuration, the analog audio bandwidth is limited to 8 kHz.

Optionally, the secondary, tertiary, and inner PIDS subcarriers may be disabled so that only the primary and outer PIDS sidebands are transmitted; this allows for an analog audio bandwidth of up to 9.4 kHz. Refer to Figure 5-3 for an illustration.

Status and control information is transmitted on reference subcarriers located on both sides of the main carrier. Each sideband has both an upper and a lower component. The PIDS logical channel is transmitted in individual subcarriers just above and below the frequency edges of the upper and lower secondary sidebands. The power level of each OFDM subcarrier is fixed relative to the other subcarriers within the same sideband. However, the absolute power levels of entire sidebands are adjustable.

Table 5-1 summarizes the spectral characteristics of the Hybrid waveform. Individual subcarriers are numbered from -81 to 81 with the center subcarrier at subcarrier number 0. Table 5-1 lists the approximate frequency ranges and bandwidths for each sideband. In Table 5-1, the subcarriers 54 to 56 and -54 to -56 are not represented. This is because they are not transmitted to avoid interference with first adjacent signals.

The *amplitude scale factors* listed in Table 5-1 and Table 5-2 refer to the multiplication constants used to scale the individual subcarriers to the appropriate power levels as defined in [9]. Refer to Section 12 for details of the subcarrier scaling operation. Refer to Subsection 6.3 for a description of the control signals that are used to select the amplitude scale factors. The resolution of the amplitude scale factors, and the actual point or points within the signal path where they are applied depends on the specific implementation and is outside the scope of this document.

For the Hybrid waveform illustrated in Figure 5-1 and Figure 5-2, all of the subcarriers within the Primary Lower sideband share a common scale factor, CH_{PL}, so that these subcarriers have the same amplitude relative to one another. Similarly, all of the subcarriers within the Primary Upper sideband share a common scale factor, CH_{PU}, so that these subcarriers have the same amplitude relative to one another. However, CH_{PL} and CH_{PU} may be different; the upper and lower sidebands may differ in average power level (asymmetric sidebands). Normally, the sideband power levels are equal, but under certain scenarios, asymmetric sidebands may be useful in order to compensate for antenna systems that do not have a symmetric frequency response or in order to reduce potential interference to another broadcast on an adjacent channel.

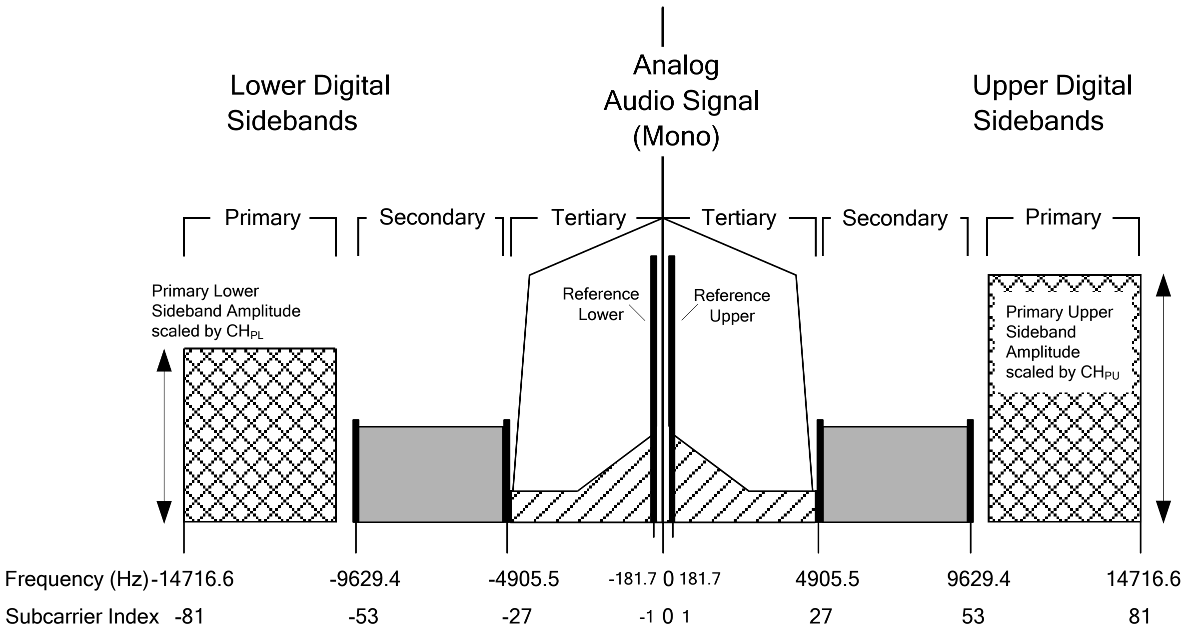

Figure 5-1: AM HD Radio Hybrid Waveform Spectrum (5 kHz Audio Configuration)

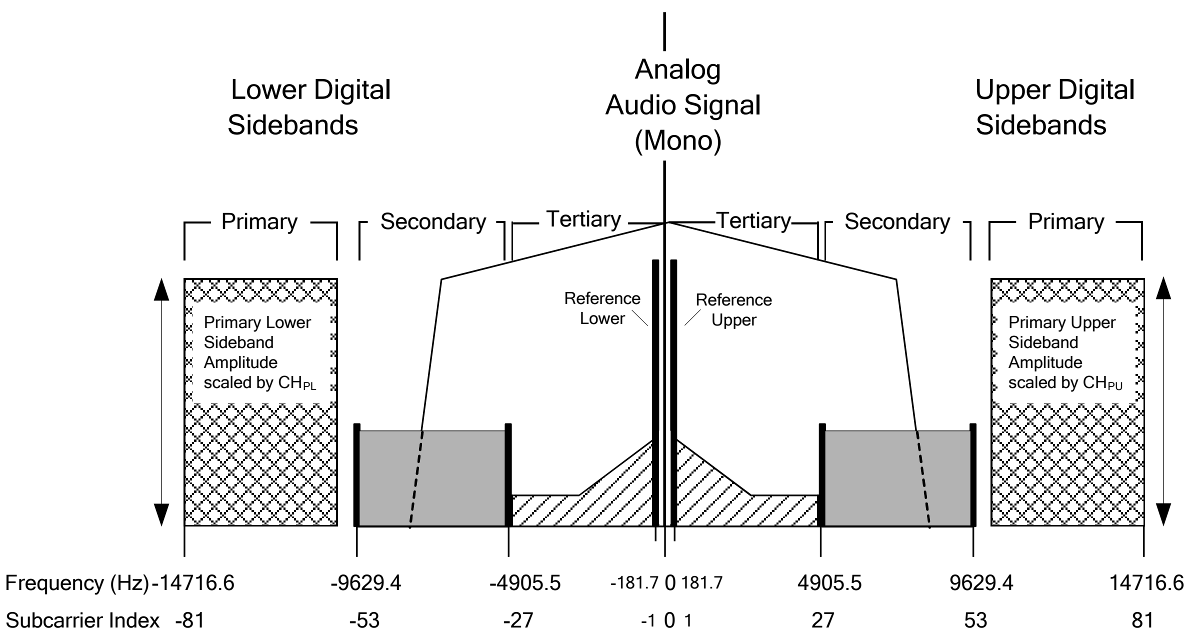

Figure 5-2: AM HD Radio Hybrid Waveform Spectrum (8 kHz Audio Configuration)

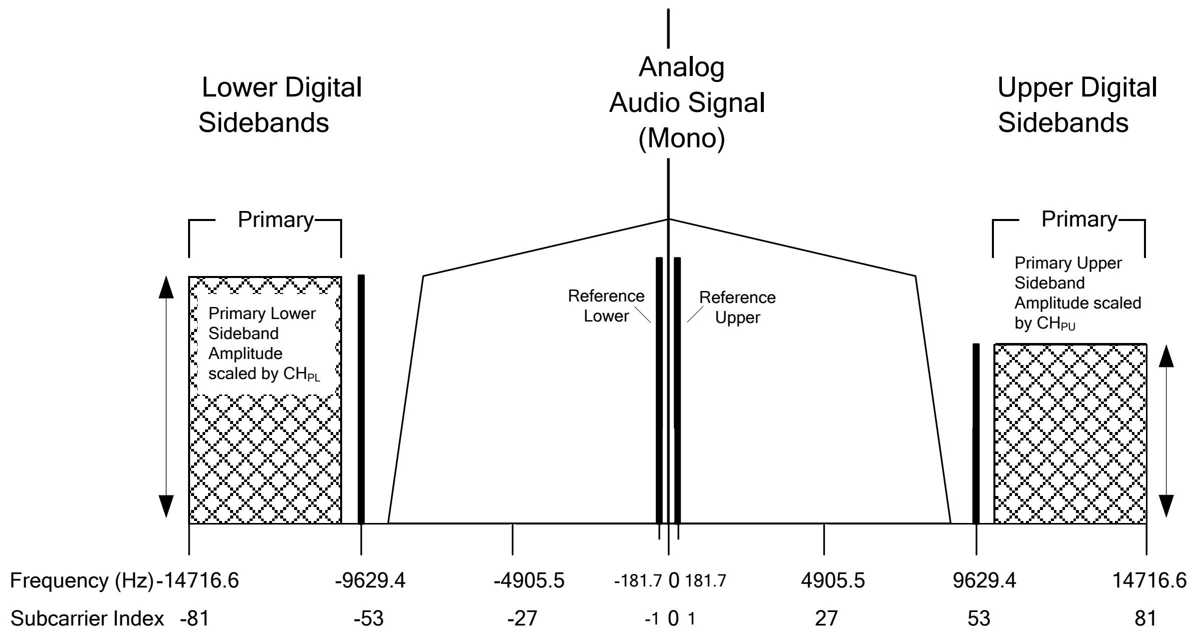

Figure 5-3: AM HD Radio Hybrid Waveform Spectrum (Reduced Digital Bandwidth Configuration)

Table 5-1: AM Hybrid Waveform Spectral Summary

Sideband	Subcarrier Range	Subcarrier Frequencies (Hz from channel center)†	Amplitude Scale Factor	Comments
Primary Upper	57 to 81	10356.1 to 14716.6	CH_{PU}	Power adjustable as defined in [9]. Primary Upper and Lower Sidebands may be adjusted independently.
Primary Lower	-57 to -81	-10356.1 to -14716.6	CH_{PL}	Power adjustable as defined in [9]. Primary Upper and Lower Sidebands may be adjusted independently.
Secondary Upper	28 to 52	5087.2 to 9447.7	CH_{S1} or CH_{S2} or 0.0	Power adjustable as defined by the Power level control setting defined in Subsection 6.3.1. Sideband may also be disabled by the control signal defined in Subsection 6.3.2.

† Subcarrier frequencies shown refer to the center frequency of the subcarrier.

Sideband	**Subcarrier Range**	**Subcarrier Frequencies (Hz from channel center)†**	**Amplitude Scale Factor**	**Comments**
Secondary Lower	-28 to -52	-5087.2 to -9447.7	CH_{S1} or CH_{s2} or 0.0	Power adjustable as defined by the Power level control setting defined in Subsection 6.3.1. Sideband may also be disabled by the control signal defined in Subsection 6.3.2.
Tertiary Upper	2 to 26	363.4 to 4723.8	$\underline{CH_{T1}}[0:24]$ or $\underline{CH_{T2}}[0:24]$ or 0.0	Each subcarrier in this sideband has a unique scale factor. Sideband may also be disabled by the control signal defined in Subsection 6.3.2.
Tertiary Lower	-2 to -26	-363.4 to -4723.8	$\underline{CH_{T1}}[0:24]$ or $\underline{CH_{T2}}[0:24]$ or 0.0	Each subcarrier in this sideband has a unique scale factor. Sideband may also be disabled by the control signal defined in Subsection 6.3.2.
Reference Upper	1	181.7	CH_B	
Reference Lower	-1	-181.7	CH_B	
PIDS1	27	4905.5	CH_{I1} or CH_{I2}	Power adjustable as defined by the Power level control setting defined in Subsection 6.3.1. Sideband may be disabled be the control signal defined in Subsection 6.3.2.
PIDS2	53	9629.4	$CH_{PU} \cdot CH_{I3}$ or $CH_{PU} \cdot CH_{I4}$ or $CH_{PU} \cdot CH_{I5}$	Power adjustable, relative to the Primary Upper sideband power, as defined by the control signal settings defined in Subsection 6.3.

Sideband	Subcarrier Range	Subcarrier Frequencies (Hz from channel center)†	Amplitude Scale Factor	Comments
PIDS1*	-27	-4905.5	CH_{I1} or CH_{I2}	* denotes complex conjugate Power adjustable as defined by the Power level control setting defined in Subsection 6.3.1. Sideband may be disabled be the control signal defined in Subsection 6.3.2.
PIDS2*	-53	-9629.4	$CH_{PL} \cdot CH_{I3}$ or $CH_{PL} \cdot CH_{I4}$ or $CH_{PL} \cdot CH_{I5}$	* denotes complex conjugate Power adjustable, relative to the Primary Lower sideband power, as defined by the control signal settings defined in Subsection 6.3.

5.4 All Digital Spectrum

In the All Digital waveform, the analog signal is replaced with higher power primary sidebands. The unmodulated AM carrier is retained. In addition, the secondary *upper sideband* moves to the higher frequencies above the primary upper sideband and the tertiary *lower sideband* moves to the lower frequencies below the primary lower sideband. The secondary lower and tertiary upper sidebands are no longer used. Furthermore, the power of both the secondary and tertiary sidebands is increased. These changes result in the overall bandwidth being reduced, making the All Digital waveform less susceptible to adjacent channel interference. The reference subcarriers are located on both sides of the unmodulated AM carrier as in the Hybrid waveform, but at a higher level. The spectrum of the All Digital waveform is illustrated in Figure 5-4.

Optionally, the secondary and tertiary subcarriers may be disabled so that only the primary and PIDS sidebands are transmitted, reducing the total transmission bandwidth to less than 9.4 kHz. Refer to Figure 5-5 for an illustration.

The power level of each of the OFDM subcarriers within a sideband is fixed relative to the unmodulated main analog carrier. Table 5-2 summarizes the spectral characteristics of the All Digital waveform.

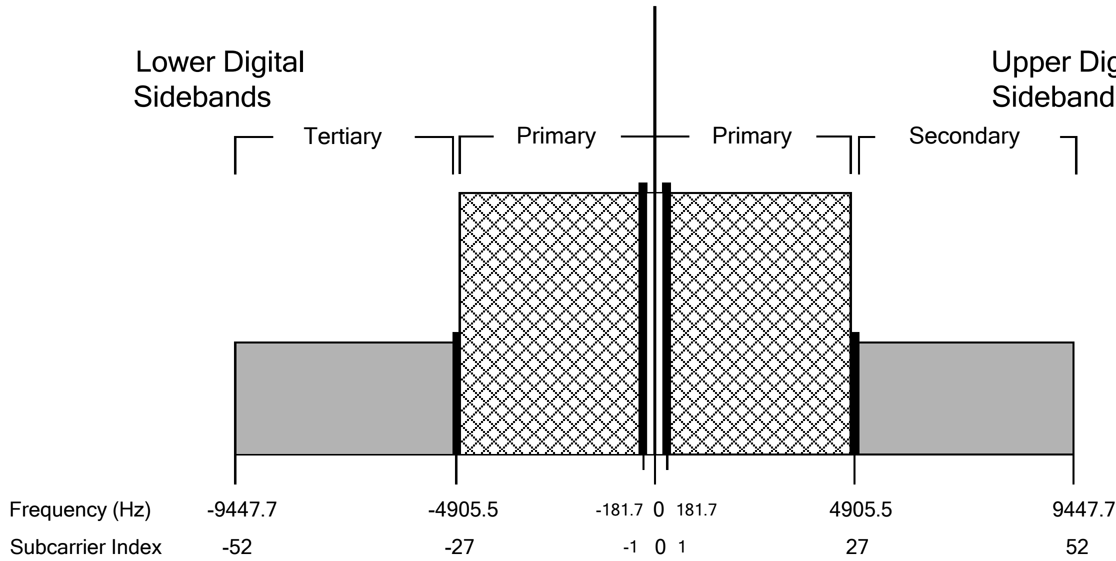

Figure 5-4: AM All Digital Waveform Spectrum

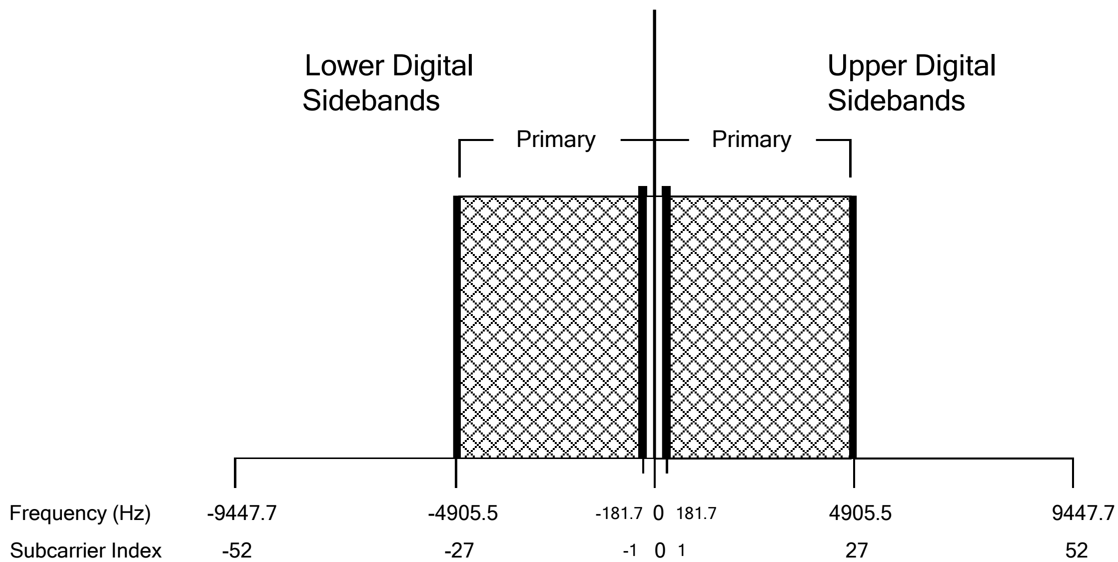

Figure 5-5: AM HD Radio All Digital Waveform Spectrum (Reduced Digital Bandwidth Configuration)

Table 5-2: AM All Digital Waveform Spectral Summary

Sideband	Subcarrier Range	Subcarrier Frequencies (Hz from channel center)‡	Amplitude Scale Factor	Comments
Primary Upper	2 to 26	363.4 to 4723.8	CD_P	
Primary Lower	-2 to -26	-363.4 to -4723.8	CD_P	
Secondary	28 to 52	5087.2 to 9447.7	CD_E or 0.0	May be disabled by the control signal defined in Subsection 6.3.2
Tertiary	-28 to -52	-5087.2 to -9447.7	CD_E or 0.0	May be disabled by the control signal defined in Subsection 6.3.2
Reference Upper	1	181.7	CD_B	
Reference Lower	-1	-181.7	CD_B	
PIDS1	27	4905.5	$CD_P \cdot CD_{I1}$ or $CD_P \cdot CD_{I2}$	Power adjustable, relative to the Primary sideband power, as defined by the control signal settings defined in Subsection 6.3.
PIDS2	-27	-4905.5	$CD_P \cdot CD_{I1}$ or $CD_P \cdot CD_{I2}$	Power adjustable, relative to the Primary sideband power, as defined by the control signal settings defined in Subsection 6.3.

‡ Subcarrier frequencies shown refer to the center frequency of the subcarrier.

6 System Control Channel

6.1 Introduction

The SCCH passes discrete transfer frames of control and status information between Layer 2, the Configuration Administrator and Layer 1. The control information passed from the Configuration Administrator to Layer 1 consists of service mode control (PSM), power level control (PL) (for Hybrid waveforms only), reduced digital bandwidth control (RDB), high-power PIDS control (HPP), and *analog audio bandwidth control* (AAB) (for Hybrid waveforms only). In addition, several bits of the system control data sequence designated "reserved" are controlled by the Configuration Administrator.

The status information passed from Layer 1 to Layer 2 consists of *absolute L1 frame number* (ALFN) and *L1 block count* (BC). Refer to Figure 6-1. This status information and the L1 block count and indicators of the state of the control information (with the exception of ALFN) are broadcast on the reference subcarriers.

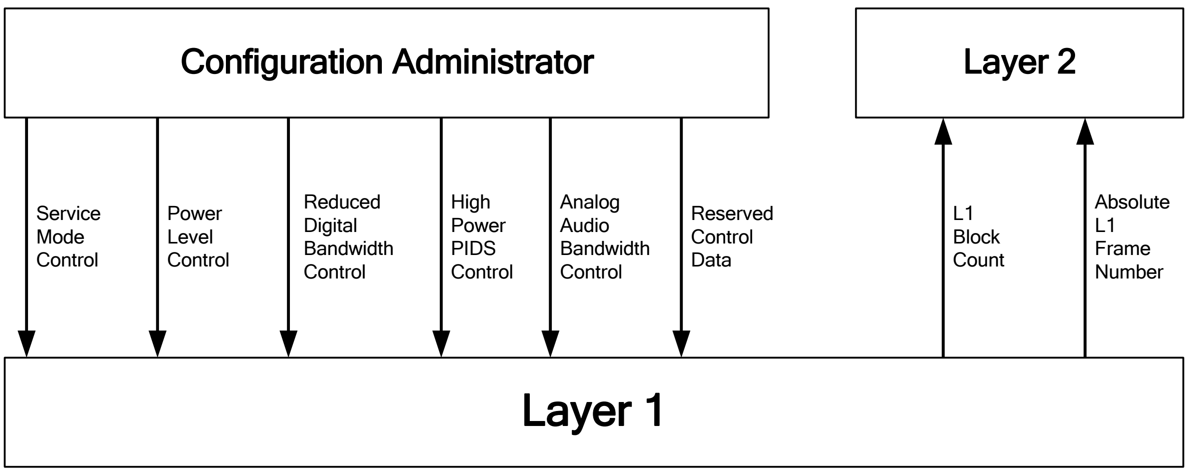

Figure 6-1: System Control Channel

The direction and rate of transfer between Layer 2, the Configuration Administrator and Layer 1 is given in Table 6-1.

Table 6-1: Transfer through the SCCH

Data	Direction	Transfer Frame Rate	Size (bits)
Service Mode Control (PSM)	Configuration Administrator ⇒ Layer 1	R_f	5
Power Level Control (PL)	Configuration Administrator ⇒ Layer 1	R_f	1
Reduced Digital Bandwidth Control (RDB)	Configuration Administrator ⇒ Layer 1	R_f	1
High-Power PIDS Control (HPP)	Configuration Administrator ⇒ Layer 1	R_f	1

Data	Direction	Transfer Frame Rate	Size (bits)
Analog Audio Bandwidth Control (AAB)	Configuration Administrator ⇒ Layer 1	R_f	1
Reserved Control Data	Configuration Administrator ⇒ Layer 1	R_f	5
L1 Block Count (BC)	Layer 1 ⇒ Layer 2	R_b	4
Absolute L1 Frame Number (ALFN)	Layer 1 ⇒ Layer 2	R_f	32

6.2 Service Mode Control

The service mode control determines the configuration and performance of the logical channels, as well as the waveform type (i.e., Hybrid or All Digital). The AM system as defined in this document supports two service modes: MA1 and MA3. MA1 is a Hybrid mode, while MA3 is an All Digital mode. Table 6-2 defines the bit assignments for the Service Mode Control.

Table 6-2: PSM Bit Assignments

Service Mode	Bit Assignment (Bits 4:0)
None	00000
MA1	00001
MA3	00010
Reserved	00011 - 11111

6.2.1 Service Mode Backward Compatibility

Service mode bit assignments greater than binary 00010 are reserved for future expansion. However, to ensure backward compatibility, all reserved service modes must maintain backward compatibility with one of the service modes MA1 and MA3. As a minimum, backward compatibility includes the PIDS logical channel, the system control data sequence (vector $\underline{R}$) conveyed over the reference subcarriers, and at least one logical channel which can support medium-quality digital audio. Refer to Table 6-3 for a definition of the default service modes that first-generation receivers will assume and with which all transmission equipment must maintain backward compatibility for all reserved service mode assignments. Any service mode that is backward compatible with Hybrid service mode MA1 (e.g., MA6, MA10, MA14,…) is also a Hybrid service mode and includes the analog AM signal.

Table 6-3: Reserved Service Modes – Defaults

Actual Service Mode	Bit Assignment (Bits 4:0)	Default Service Mode	Backward Compatible Logical Channels/Elements	Logical Channels Free to be Redefined
MA4	000 11	Reserved	N/A	N/A
MA5	001 00	Reserved	N/A	N/A
MA6	001 01	MA1	P1, PIDS, $\underline{R}$, Analog	P3

Actual Service Mode	Bit Assignment (Bits 4:0)	Default Service Mode	Backward Compatible Logical Channels/Elements	Logical Channels Free to be Redefined
MA7	001 10	MA3	P1, PIDS, R	P3
MA8	001 11	Reserved	N/A	N/A
MA9	010 00	Reserved	N/A	N/A
MA10	010 01	MA1	P1, PIDS, R, Analog	P3
MA11	010 10	MA3	P1, PIDS, R	P3
MA12	010 11	Reserved	N/A	N/A
MA13	011 00	Reserved	N/A	N/A
MA14	011 01	MA1	P1, PIDS, R, Analog	P3
MA15	011 10	MA3	P1, PIDS, R	P3
MA16	011 11	Reserved	N/A	N/A
MA17	100 00	Reserved	N/A	N/A
MA18	100 01	MA1	P1, PIDS, R, Analog	P3
MA19	100 10	MA3	P1, PIDS, R	P3
MA20	100 11	Reserved	N/A	N/A
MA21	101 00	Reserved	N/A	N/A
MA22	101 01	MA1	P1, PIDS, R, Analog	P3
MA23	101 10	MA3	P1, PIDS, R	P3
MA24	101 11	Reserved	N/A	N/A
MA25	101 00	Reserved	N/A	N/A
MA26	110 01	MA1	P1, PIDS, R, Analog	P3
MA27	110 10	MA3	P1, PIDS, R	P3
MA28	110 11	Reserved	N/A	N/A
MA29	111 00	Reserved	N/A	N/A
MA30	111 01	MA1	P1, PIDS, R, Analog	P3
MA31	111 10	MA3	P1, PIDS, R	P3
MA32	111 11	Reserved	N/A	N/A

6.2.2 Service Mode Switching

The service mode control is received from the Configuration Administrator via the SCCH at the rate R_f. Service mode changes are invoked only on an *L1 frame* boundary (see Subsection 6.3).

6.3 Subcarrier Scaling Control Signals

The system provides three control signals, RDB, HPP, and PL that determine the power profile of the various sidebands. These three controls are received from the Configuration Administrator at the *L1 frame rate*, R_f, and any change can be effected directly at an L1 frame boundary upon receipt without interrupting service. Table 6-4 and Table 6-5 show the amplitude scale factor selections versus the states of the subcarrier scaling control signals for service mdoes MA1 and MA3, respectively.

Refer to Subsections 6.3.1, 6.3.2, and 6.3.3 for detailed descriptions of the RDB, HPP, and PL control signals, respectively. Each of these control signals has a corresponding indicator that is broadcast in the system control data sequence, as described in Subsection 11.2. Refer to [9] for details on the actual power levels associated with the amplitude scale factors shown in Table 6-4 and Table 6-5.

Table 6-4: Subcarrier Scaling Control Signal States for Service Mode MA1

Subcarrier Scaling Control Signal State			Amplitude Scale Factor Selection				Comments
RDB	HPP	PL	PIDS2 / PIDS2*	PIDS1 / PIDS1*	Secon-dary	Tertiary	
0	0	0	$CH_{PU} \cdot CH_{I3}$ $CH_{PL} \cdot CH_{I3}$	CH_{I1}	CH_{S1}	$\underline{CH_{T1}}[0{:}24]$	Standard Power Profile: Low-power Secondary/Tertiary/PIDS1 Low-power PIDS2
0	0	1	$CH_{PU} \cdot CH_{I4}$ $CH_{PL} \cdot CH_{I4}$	CH_{I2}	CH_{S2}	$\underline{CH_{T2}}[0{:}24]$	High-power Secondary/Tertiary/PIDS1 Medium-power PIDS2
0	1	0	$CH_{PU} \cdot CH_{I5}$ $CH_{PL} \cdot CH_{I5}$	CH_{I1}	CH_{S1}	$\underline{CH_{T1}}[0{:}24]$	Low-power Secondary/Tertiary/PIDS1 High Power PIDS2
0	1	1	$CH_{PU} \cdot CH_{I5}$ $CH_{PL} \cdot CH_{I5}$	CH_{I2}	CH_{S2}	$\underline{CH_{T2}}[0{:}24]$	High-power Secondary/Tertiary/PIDS1 High Power PIDS2
1	X	X	$CH_{PU} \cdot CH_{I5}$ $CH_{PL} \cdot CH_{I5}$	0.0	0.0	0.0	Secondary/Tertiary/PIDS1 are disabled High Power PIDS2

Table 6-5: Subcarrier Scaling Control Signal States for Service Mode MA3

Subcarrier Scaling Control Signal State			Amplitude Scale Factor Selection		Comments
RDB	HPP	PL	PIDS2 / PIDS1	Secondary / Tertiary	
0	0	X	$CD_P \cdot CD_{I1}$	CD_E	Standard Power Profile
0	1	X	$CD_P \cdot CD_{I2}$	CD_E	High-Power PIDS
1	X	X	$CD_P \cdot CD_{I2}$	0.0	Secondary/Tertiary Sidebands Disabled High Power PIDS

6.3.1 Power Level Control

In the Hybrid waveform, the nominal level of the secondary, PIDS1, and tertiary sidebands is one of two selectable values: low or high. Power level control (PL) specifies which level is to be employed. When PL is a logical 0 (low power level), the Hybrid subcarriers are scaled by CH_{S1}, CH_{I1}, and CH_{T1} to increase digital coverage. When PL is a logical 1 (high power level), the Hybrid subcarriers are scaled by CH_{S2}, CH_{I2}, and CH_{T2} to reduce analog interference. Refer to [9] for details. When transmitting the All Digital waveform, PL is ignored.

When RDB is set to 1, the state of the PL control signal is ignored.

6.3.2 Reduced Digital Bandwidth Control

The *Reduced Digital Bandwidth Control* (RDB) is used to limit the spectrum of the MA1 and MA3 waveforms. In service mode MA1, RDB may be used to reduce interference to the analog host and accommodate an analog bandwidth of up to 9.4 kHz. In service mode MA3, RDB may be used to reduce interference to adjacent channels.

When RDB is 0 (standard configuration), the secondary and tertiary sidebands are enabled, and the PIDS subcarriers are all active.

When RDB is 1 (reduced bandwidth configuration), the secondary and tertiary sidebands are disabled by setting the appropriate amplitude scale factors to zero. In addition, for service mode MA1, the PIDS1 and PIDS1* subcarriers are disabled. In both service modes MA1 and MA3, setting RDB to 1 will set the active PIDs subcarriers to the high-power configuration.

When RDB is set to 1, the state of the PL and HPP control signals is ignored.

6.3.3 High-Power PIDS Control

The High Power PIDS Control (HPP) is used to set the level of the PIDS2/PIDS2* subcarriers in service mode MA1 and the PIDS1/PIDS2 subcarriers in service mode MA3. If HPP is 0, these PIDS subcarriers will be in the standard low-power configuration. If HPP is 1, these PIDS subcarriers will be in the high-power configuration.

When RDB is set to 1, the state of the HPP control signal is ignored.

6.4 Analog Audio Bandwidth Control

In the standard configuration (RDB = 0) of the hybrid waveform, the bandwidth of the analog audio is one of two selectable values: 5 kHz or 8 kHz. Analog audio bandwidth control (AAB) specifies which bandwidth is to be employed. When AAB is a logical 0, the bandwidth indicated is 5 kHz. When AAB is a logical 1, the bandwidth indicated is 8 kHz. When transmitting the All Digital waveform, it is ignored. Analog audio bandwidth control is received from the Configuration Administrator at the L1 frame rate, R_f, and any change can be effected directly (at an L1 frame boundary) upon receipt without interrupting service. Digital coverage of a Hybrid station (primary subcarriers only) is adversely impacted by a second adjacent Hybrid transmission with 8 kHz audio bandwidth.

In the reduced digital bandwidth configuration (RDB = 1) of the hybrid waveform, the state of AAB is not applicable and shall be set to a value of zero.

6.5 Absolute L1 Frame Number

The transmitted HD Radio signal may be regarded as a series of unique L1 frames of duration T_f. In order to reference all transmissions to absolute time, each L1 frame is associated with an ALFN. This universal frame numbering scheme assumes that the start of ALFN 0 occurred at the GPS epoch -00:00:00 Coordinated Universal Time (UTC) on January 6, 1980. The start of every subsequent L1 frame occurs at an exact integer multiple of T_f after that instant in time. A new GPS epoch starts every 1024 weeks. The current ALFN is a binary number determined by subtracting the GPS start time (00:00:00 on January 6, 1980) from the current GPS time (making allowance for the GPS epoch), expressing the difference in seconds, and multiplying the result by the frame rate, R_f.

The ALFN, which is passed to Layer 2 via the SCCH at the rate R_f, is used to schedule the delivery of time-critical programming. It is not broadcast as part of the transmitted HD Radio signal.

6.6 L1 Block Count

Each L1 frame may be considered to consist of eight *L1 blocks* of duration T_b. The BC indicates the position of the current L1 block within the L1 frame. An L1 block count of 0 signifies the start of an L1 frame, while a BC of 7 designates the final L1 block in an L1 frame. Table 6-6 defines the bit assignments for the L1 Block Count passed to L2.

The three least significant bits of BC are passed to Layer 2 via the SCCH at the rate R_b. They are broadcast on the reference subcarriers and are used by the receiver to aid in synchronization. The most significant bit of BC is not used. An illustration of the relationship of L1 blocks to L1 frames is shown in Figure 6-2.

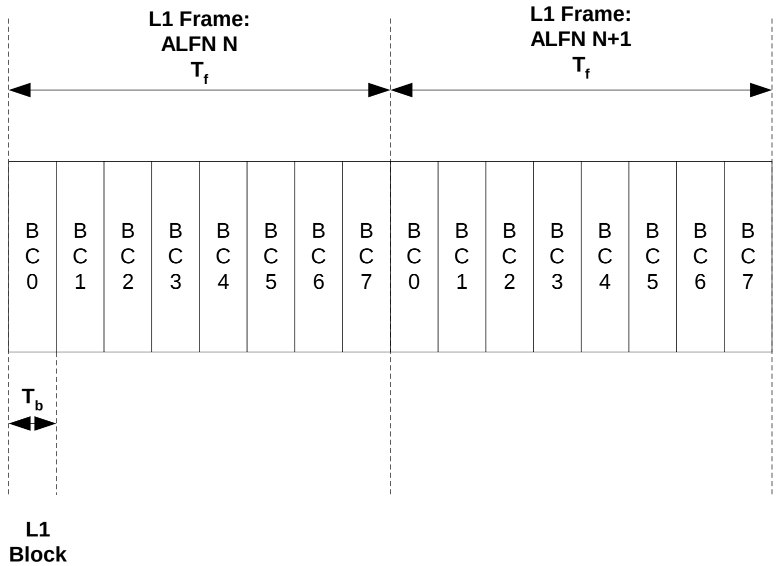

Figure 6-2: L1 Frames and Blocks

Table 6-6: BC Bit Assignments

L1 Block Count	BC Bit Assignment (Bits 3:0)
0	0000
1	0001
2	0010
3	0011
4	0100
5	0101
6	0110
7	0111
Not Used	1000 - 1111

7 Logical Channels

7.1 Introduction

A logical channel is a signal path that conducts L2 PDUs through Layer 1 with a specified grade of service. The available logical channels are P1, P3, and PIDS. Logical channels are defined by their characterization parameters and configured by the service mode. They are used in both the Hybrid and All Digital waveforms.

7.2 Characterization Parameters

For a given service mode, the grade of service of a particular logical channel may be uniquely quantified using three characterization parameters: transfer, latency, and robustness. Channel code rate, interleaver depth, digital diversity delay, and spectral mapping are the determinants of the characterization parameters.

7.2.1 Transfer

The throughput of a logical channel is called transfer. The block-oriented operations of Layer 1 (such as interleaving) require that data be processed in discrete transfer frames rather than continuous streams. As a result, throughput is defined in terms of *transfer frame size* (in bits) and *transfer frame rate* (in Hz, or the number of transfer frames per second). This Layer 1 framing effectively defines the alignment of L2 PDUs.

Each transfer frame is uniquely identified by its *transfer frame number* $F^{n}_{m1:m2}$, where n is the ALFN with which the transfer frame is associated and m1:m2 is the *BC range* that is spanned by the transfer frame within the L1 frame n. Thus, the BC range indicates the position of the transfer frame within the L1 frame. In cases where a transfer frame is split and delayed in L1 with the result that it is transmitted in two different ALFNs, the superscript n refers to the first instance of its transmission. The transfer frame number is not broadcast as part of the transmitted HD Radio signal.

All transfer frames are conducted through Layer 1 at one of two rates:

- the L1 frame rate, $R_f = \dfrac{1}{T_f}$
- the *L1 block rate*, $R_b = \dfrac{1}{T_b}$

The transfer frame rate relationships are summarized in Table 7-1 and illustrated in Figure 7-1.

Table 7-1: Transfer Frame Rate Relationships

Transfer Frame		
Type	**Duration (Seconds)**	**Rate (Hz)**
L1 block	T_b	$R_b = 8 \cdot R_f$
L1 frame	T_f	R_f

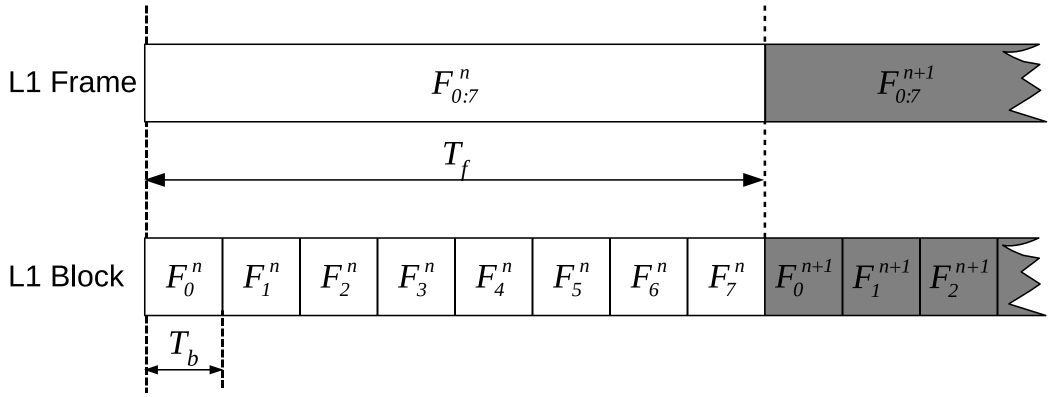

Figure 7-1: Transfer Frame Timing Relationships

Spectral mapping and channel code rate determine the transfer of a logical channel, since spectral mapping limits capacity and coding overhead limits information throughput. Interleaver depth is also a factor, because transfer frames are conducted through Layer 1 at rates corresponding to the interleaver depth of their logical channel.

7.2.2 Latency

Latency is the delay that a logical channel imposes on a transfer frame as it traverses Layer 1. The latency of a logical channel is defined as the sum of its interleaver depth and digital diversity delay. It does not include processing delays in Layer 1 nor does it include delays imposed in upper layers.

The interleaver depth determines the amount of delay imposed on a logical channel by an interleaver. The AM HD Radio system employs two interleaver depths: L1 block and L1 frame. A digital diversity delay of T_{dd} is also employed on some logical channels, such as P1 in service mode MA1, for example.

Higher layers assign information to logical channels with the requisite latency through service mode selection. Three latencies are specified for the system, as defined in Table 7-2.

Table 7-2: Latency Summary

Description	Delay
L1 block	T_b
L1 frame	T_f
L1 frame plus digital diversity delay	$T_f + T_{dd}$

7.2.3 Robustness

Robustness is the ability of a logical channel to withstand channel impairments such as noise, interference, and *grounded conductive structures* (GCS). There are eight relative levels of robustness designed into Layer 1 of the AM air interface. A robustness of 1 indicates a very high level of resistance to channel impairments, while a robustness of 8 indicates a lower tolerance for channel-induced errors. As with latency, higher layers must determine the required robustness of a logical channel before selecting a service mode.

Spectral mapping, channel code rate, interleaver depth, the power level of each sideband, and digital diversity delay determine the robustness of a logical channel. Spectral mapping affects robustness by setting the relative power level, spectral interference protection, and frequency diversity of a logical channel. Channel coding increases robustness by introducing redundancy into the logical channel. Interleaver depth influences performance in *fading*, thereby affecting the robustness of the logical channel. Reducing the power of one or both primary sidebands will have a commensurate impact on robustness.

Finally, some logical channels in certain service modes delay transfer frames by a fixed duration to realize time diversity. This digital diversity delay also affects robustness, since it mitigates the effects of the mobile radio channel.

7.2.4 Assignment of Characterization Parameters

Table 7-3 and Table 7-4 show the characterization parameters of each logical channel for every service mode. Transfer is presented in terms of transfer frame size and transfer frame rate. The relative robustness figures are approximate. Exact performance may vary depending on the specific channel conditions as well as individual sideband power. The robustness figures provided in the following tables assume equal power levels in each primary sideband.

Table 7-3: Logical Channel Characterization – Service Mode MA1

Logical Channel	Transfer			Relative Robustness
	Frame Size (Bits)	Frame Rate (Hz)	L1 Latency (s)	
P1	3750	R_b	$T_f + T_{dd}$	5
P3	24000	R_f	T_f	6 (PL=High) or 8 (PL=Low)
PIDS	80	R_b	T_b	3 (PL=High) or 7 (PL=Low)

Table 7-4: Logical Channel Characterization – Service Mode MA3

Logical Channel	Transfer			Relative Robustness
	Frame Size (Bits)	Frame Rate (Hz)	L1 Latency (s)	
P1	3750	R_b	$T_f + T_{dd}$	1
P3	30000	R_f	$T_f + T_{dd}$	4
PIDS	80	R_b	T_b	2

Information throughput of a logical channel can be calculated using these tables and the following formula:

throughput (bits/s) = transfer frame size (bits) · transfer frame rate (Hz)

For example, in service mode MA1, the throughput for logical channel P1 is calculated as follows:

throughput = 3750 · (8 · 44100/65536) ≈ 20.2 kbits/s

7.3 Spectral Mapping

For a given service mode, each logical channel is applied to a frequency sideband. Figure 7-2 through Figure 7-5 show the spectral mapping for each logical channel for every service mode. In these figures, the annotated frequencies represent offsets from the channel center frequency.

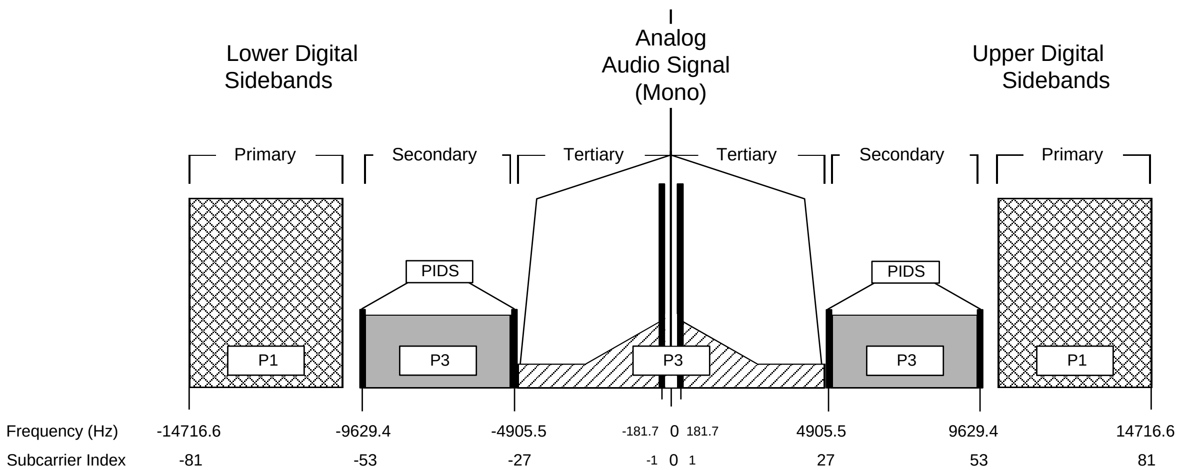

Figure 7-2: Logical Channel Spectral Mapping – Service Mode MA1 (5 kHz Audio Configuration)

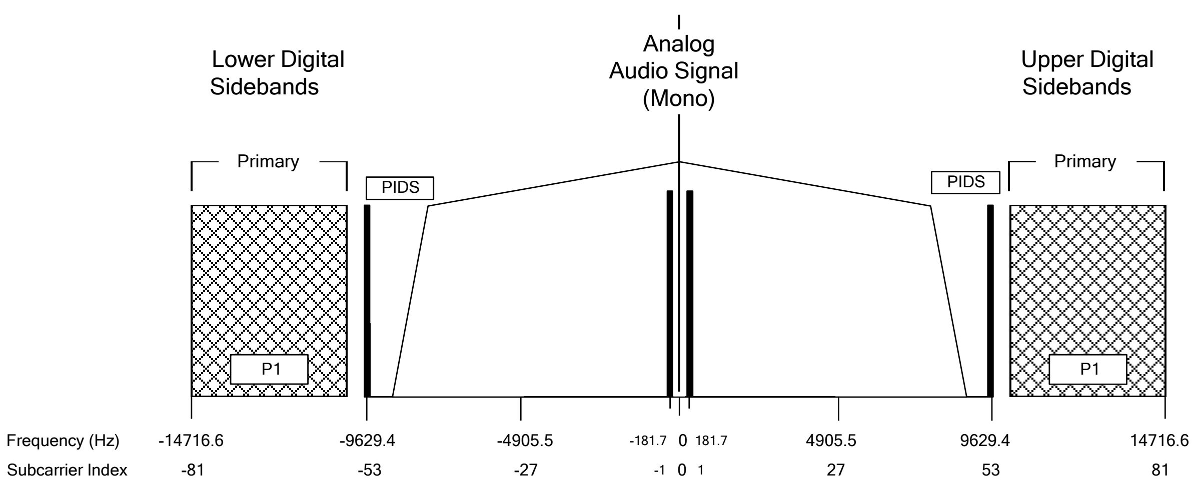

Figure 7-3: Logical Channel Spectral Mapping – Service Mode MA1 (Reduced Digital Bandwidth Configuration)

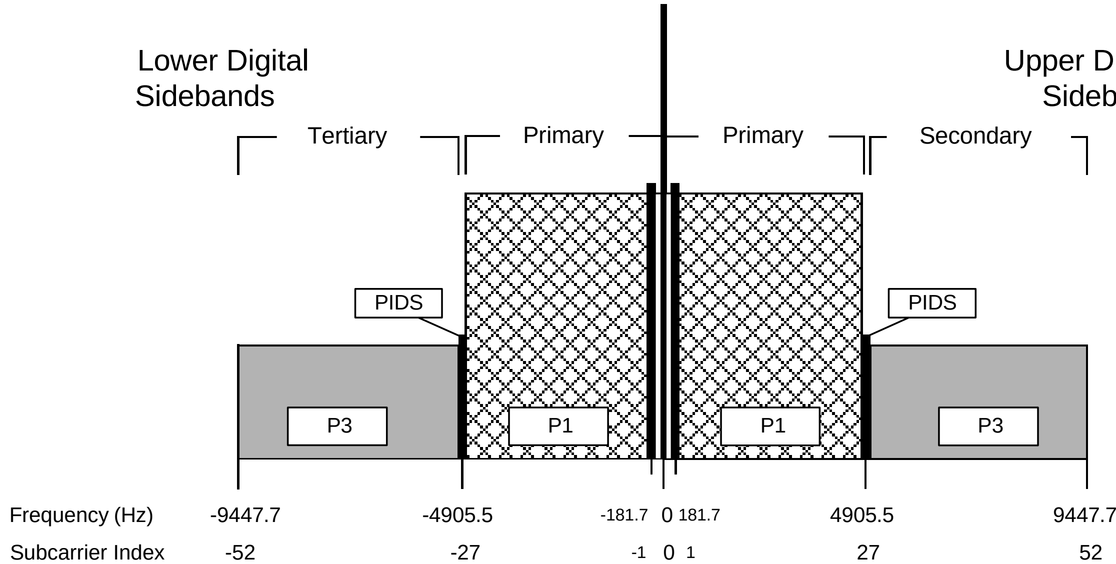

Figure 7-4: Logical Channel Spectral Mapping – Service Mode MA3

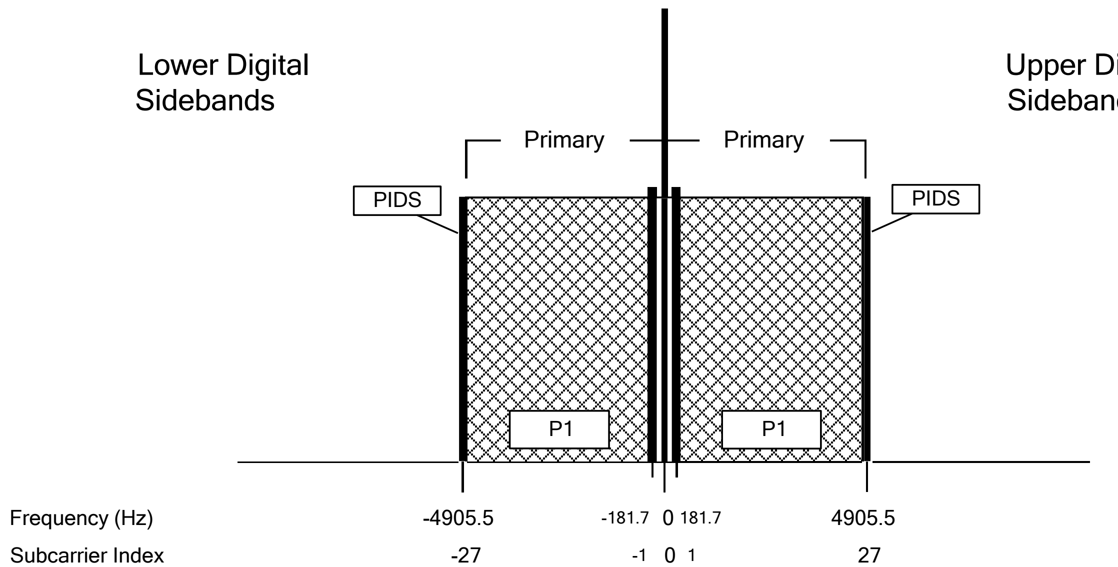

Figure 7-5: Logical Channel Spectral Mapping – Service Mode MA3 (Reduced Digital Bandwidth Configuration)

7.4 Framing and Synchronization

The logical channels share a common, absolute time reference so that all transfer frames are precisely aligned. As described in Subsection 7.2.1, each transfer frame is assigned a unique transfer frame number $F^{n}_{m1:m2}$, where n is the ALFN, and m1:m2 is the BC range that designates the position of the transfer frame within the indexed L1 frame. This numbering scheme allows all transfer frames to be referenced to an absolute transmission time. Further details of system timing alignment are given in Section 14.

8 Scrambling

The bits in each logical channel are scrambled to randomize the time-domain data and aid in receiver synchronization. As shown in Figure 8-1, there are three parallel scramblers, one for each logical channel.

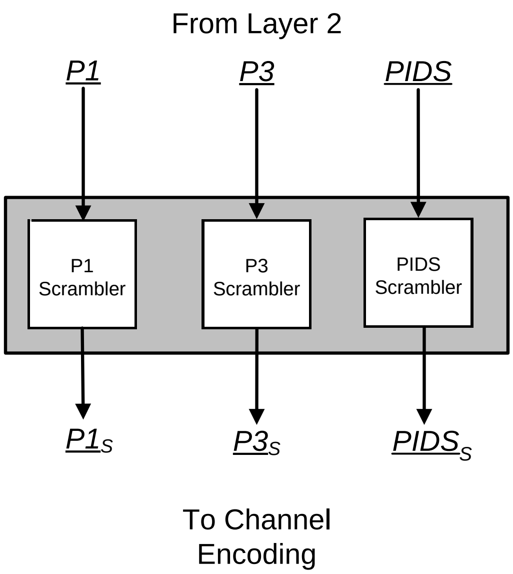

Figure 8-1: Scrambling Functional Block Diagram

The inputs to the scramblers are the active logical channels from Layer 2, as selected by the service mode control. These inputs are delivered in discrete transfer frames whose size and rate are defined in Table 7-3 and Table 7-4 for a given service mode. The outputs of the scramblers are transfer frames of scrambled bits for each of the active logical channels. These transfer frames are passed to the channel encoding process for forward error correction.

8.1 Scrambler Operation

All parallel scramblers are identical but operate at different rates, depending upon the active service mode. A detailed block diagram of the scrambler is shown in Figure 8-2. Each scrambler generates a maximal-length scrambling sequence using a linear feedback shift register with the following primitive polynomial: $P(x) = 1 \oplus x^2 \oplus x^{11}$. A given bit of a scrambled transfer frame is generated by modulo-2 adding the associated input bit with the corresponding bit of the scrambling sequence.

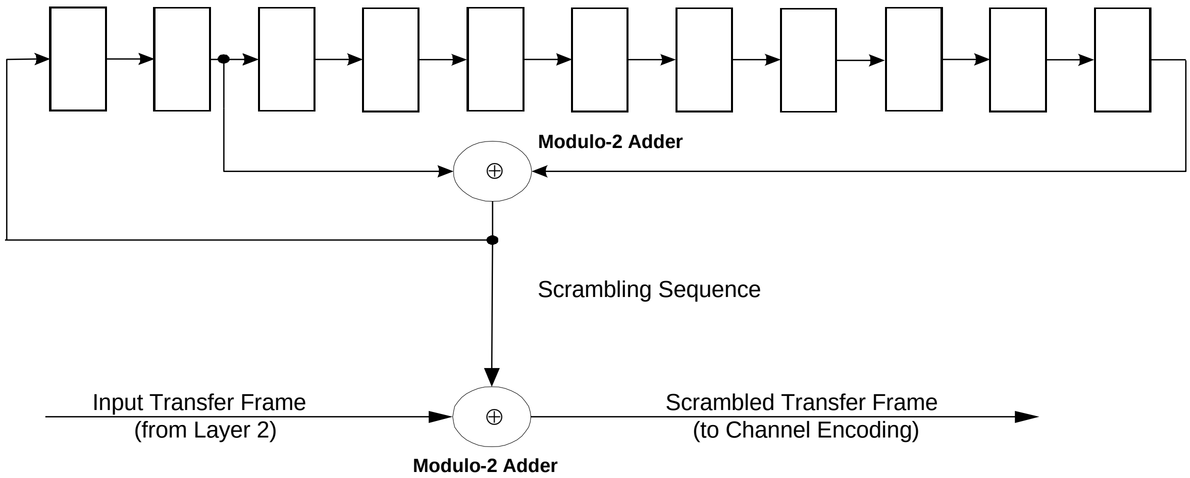

Figure 8-2: Scrambler Block Diagram

For each logical channel, the scrambler is reset to state 0111 1111 111 upon receipt of a new transfer frame. The first bit of a scrambled transfer frame is generated by modulo-2 adding the first bit of the input transfer frame with the scrambling bit generated when the shift register is set to the initial state. The process then continues until the last bit of the input transfer frame is scrambled.

9 Channel Encoding

Channel encoding improves system performance by increasing the robustness of the signal in the presence of interference and channel impairments. As shown in Figure 9-1, the channel encoding process is characterized by the single operation of convolutional encoding.

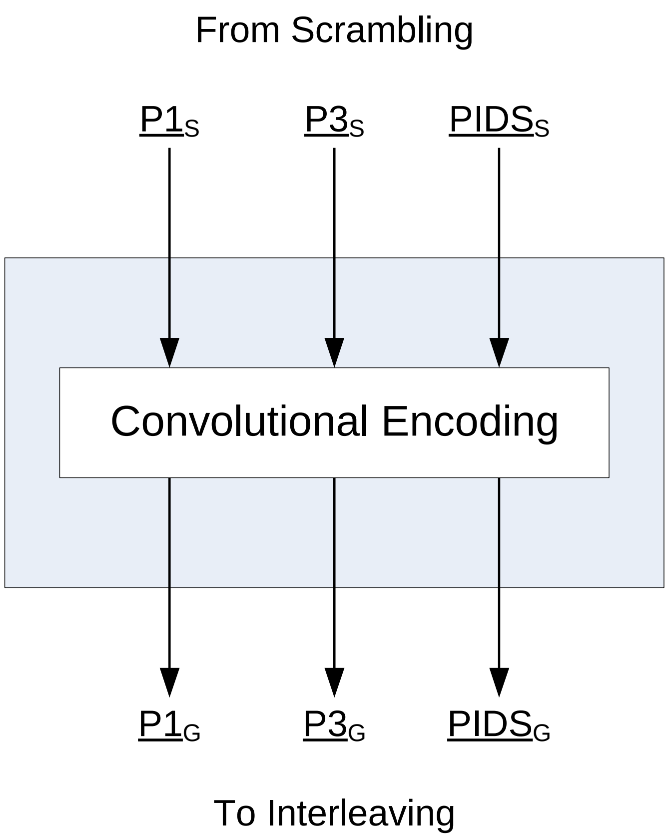

Figure 9-1: Channel Encoding Functional Block Diagram

The inputs to the channel encoding process are transfer frames of scrambled bits carried through the active logical channels. The size and rate of transfer are defined in Table 7-3 and Table 7-4 for a given service mode. The outputs of the channel encoding process are transfer frames of encoded bits associated with each of the active logical channels and are passed to interleaving.

In the ensuing subsections, for notational convenience, the logical channel vectors at a particular stage of processing are represented in shorthand notation by their subscript. For example, the scrambled inputs $\underline{P1}_S$, $\underline{P3}_S$, and $\underline{PIDS}_S$ are represented by $\underline{S}$, while the encoded outputs $\underline{P1}_G$, $\underline{P3}_G$, and $\underline{PIDS}_G$ are represented by $\underline{G}$.

9.1 Convolutional Encoding

Convolutional encoding consists of three primary operations: *mother code* generation, *puncturing*, and parallel-to-serial conversion. Each of these operations is described below.

9.1.1 Mother Code Generation

The convolutional encoders associated with each logical channel employ select generator polynomials to form a rate 1/3 mother code. Each convolutional encoder outputs 3 encoded bits $g_{h,i}$ for every input bit s_i, i=0,1,2,…,N-1, creating a codeword matrix $\underline{\underline{G}}$ of dimension 3 x N:

$$\underline{\underline{G}} = \begin{bmatrix} g_{1,0} & g_{1,1} & \cdots & g_{1,N-1} \\ g_{2,0} & g_{2,1} & \cdots & g_{2,N-1} \\ g_{3,0} & g_{3,1} & \cdots & g_{3,N-1} \end{bmatrix}$$

where N is the length of $\underline{S}$, and h=1,2,3 indexes the codeword bits for a given input information bit. Each column of $\underline{\underline{G}}$ represents the encoded output for a given input bit.

9.1.2 Puncturing

Most service modes require puncturing of the mother codeword to produce a slightly higher code rate, thereby allowing a higher information rate through the same physical bandwidth. The codeword matrix $\underline{\underline{G}}$ is punctured over a puncture period P. For every P encoded bits, certain bits $g_{h,i}$ are not transmitted. A puncture matrix spanning the encoded bits over a puncture period defines which encoded bits are transmitted. A puncture pattern is formed by repeating the puncture matrix over all information bits in the transfer frame.

9.1.3 Parallel-to-Serial Conversion

After the mother code bits are appropriately punctured, the parallel-to-serial converter multiplexes them by concatenating the columns of $\underline{\underline{G}}$ into a single vector $\underline{G}$.

9.1.4 Convolutional Encoders

Table 9-1 presents the three code rates used in the AM system, along with their associated puncture matrices. The codes are designated E1, E2, and E3. The last 8 bits of a given transfer frame are used to initialize the delay elements of the corresponding convolutional encoder for that transfer frame.

Table 9-1: AM Convolutional Codes

Encoder	Punctured Code Rate	Puncture Matrix	Mother Code Rate
E1	5/12	$\begin{bmatrix} 1 & 1 & 1 & 1 & 1 \\ 0 & 0 & 0 & 1 & 1 \\ 1 & 1 & 1 & 1 & 1 \end{bmatrix}$	1/3
E2	2/3	$\begin{bmatrix} 1 & 1 & 1 & 1 \\ 0 & 0 & 0 & 0 \\ 1 & 0 & 1 & 0 \end{bmatrix}$	1/3

E3	1/3	$\begin{bmatrix}1\\1\\1\end{bmatrix}$	1/3

9.1.4.1 E1 Convolutional Encoder

The E1 convolutional encoder, illustrated in Figure 9-2, uses a constraint length 9 rate 1/3 mother code punctured to rate 5/12. The generator polynomial used is represented in Table 9-2 below. This code is represented in octal format.

Table 9-2: E1 Convolutional Encoder Generator Polynomials – Rate 1/3 Mother Code

First Generator	Second Generator	Third Generator
561	657	711

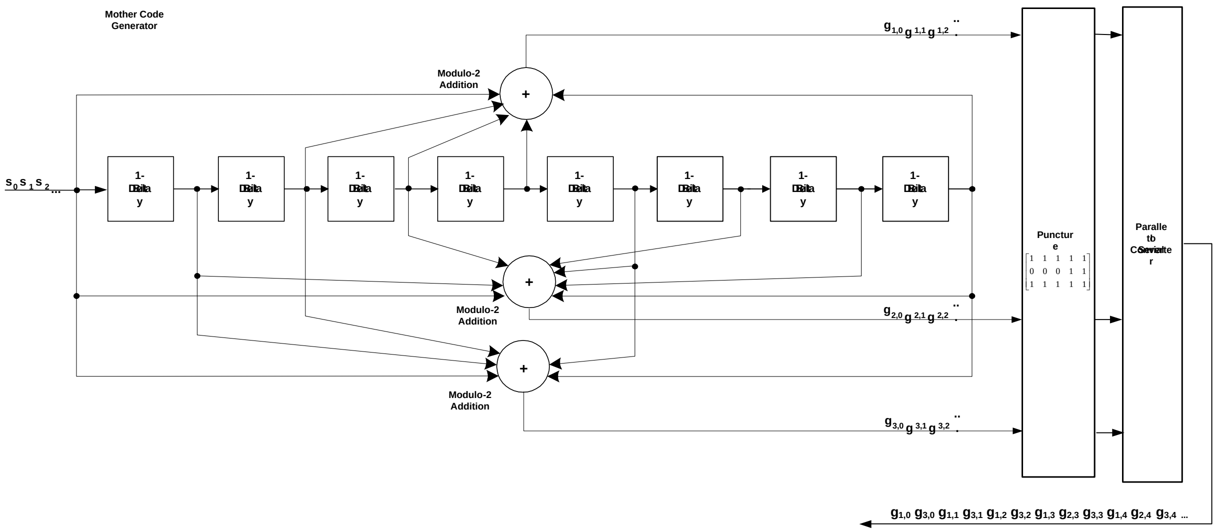

Figure 9-2: E1 Convolutional Encoder

9.1.4.2 E2 Convolutional Encoder

The E2 convolutional encoder, illustrated in Figure 9-3, uses a constraint length 9 rate 1/3 mother code punctured to rate 2/3. The generator polynomial used is represented in Table 9-3. This code is represented in octal format.

Table 9-3: E2 Convolutional Encoder Generator Polynomials

First Generator	Second Generator	Third Generator
561	753	711

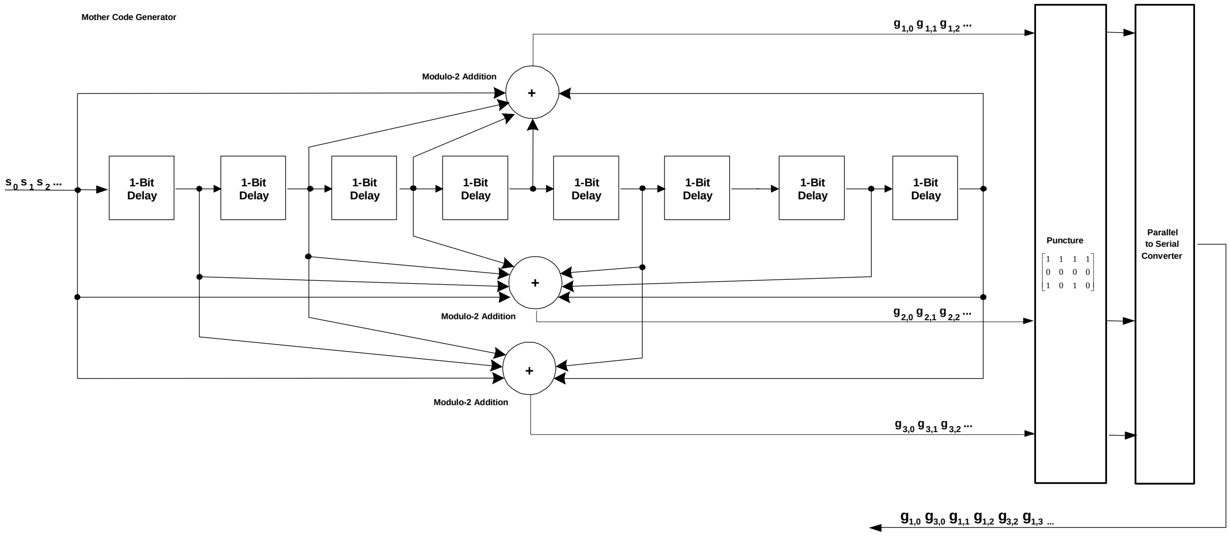

Figure 9-3: E2 Convolutional Encoder

9.1.4.3 E3 Convolutional Encoder

The E3 convolutional encoder, illustrated in Figure 9-4, uses a constraint length 9 rate 1/3 un-punctured mother code. The general polynomial used is represented in octal format as shown in Table 9-4.

Table 9-4: E3 Convolutional Encoder Generator Polynomials

First Generator	Second Generator	Third Generator
561	753	711

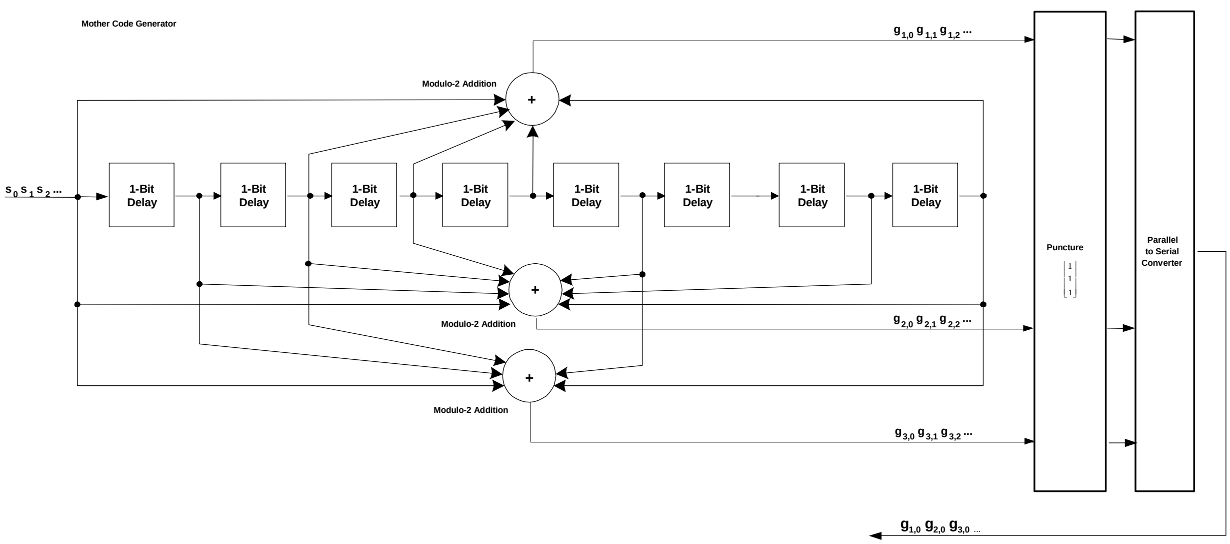

Figure 9-4: E3 Convolutional Encoder

9.2 Channel Encoding Data Flow

The channel encoding process for each logical channel in each service mode is specified in Subsection 9.2.1 through Subsection 9.2.2.

9.2.1 Service Mode MA1

The P1, P3, and PIDS logical channels are active in service mode MA1. The flow of their transfer frames through the channel encoding process is shown in Figure 9-5.

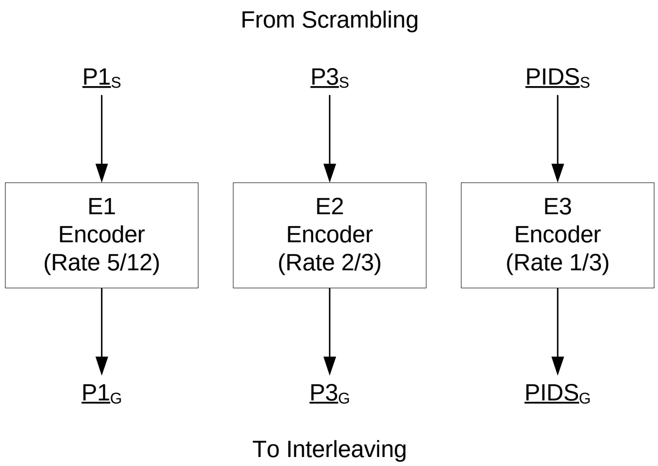

Figure 9-5: Channel Encoding – Service Mode MA1

9.2.2 Service Mode MA3

The P1, P3, and PIDS logical channels are active in service mode MA3. The flow of their transfer frames through the channel encoding process is shown in Figure 9-6.

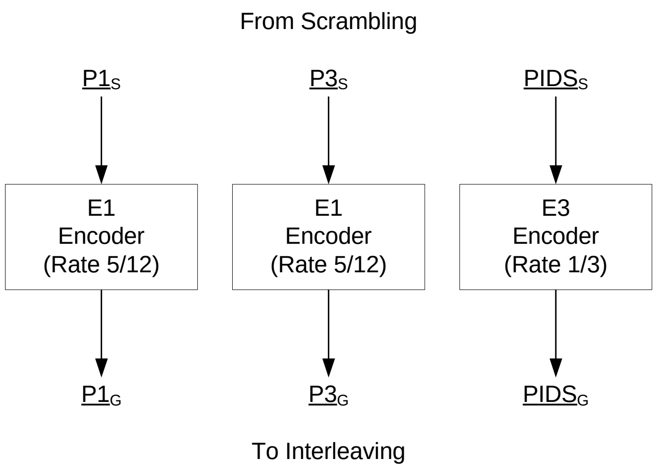

Figure 9-6: Channel Encoding – Service Mode MA3

10 Interleaving

10.1 Introduction

Interleaving provides both time and frequency diversity. As shown in Figure 10-1, interleaving for the AM system is characterized by four primary operations: *subframe generation*, digital *diversity delay*, *transmit time alignment*, and *bit mapping*.

Subframe generation is the process of accumulating the bits of one or more transfer frames and splitting them into subframes as an initial step in assigning data to the various interleaver matrices. Digital diversity delay is the process of adding delay to a select group of subframes for the purpose of adding additional time diversity beyond that provided by the interleaver matrices. *Transmit time alignment*, T_{T1a}, is associated with the P1 logical channel and allows more flexibility in the transfer of this data. It is adjusted to provide a digital diversity delay between main and backup streams to achieve precisely T_{dd}. The bit mapping process assigns each subframe bit to a row and column in an interleaver matrix as well as to a unique bit position in the digital word within the interleaver matrix element.

The manner in which the logical channels are split into subframes, delayed and mapped into interleaver matrices, is shown in Figure 10-4 through Figure 10-6 for each logical channel in each service mode.

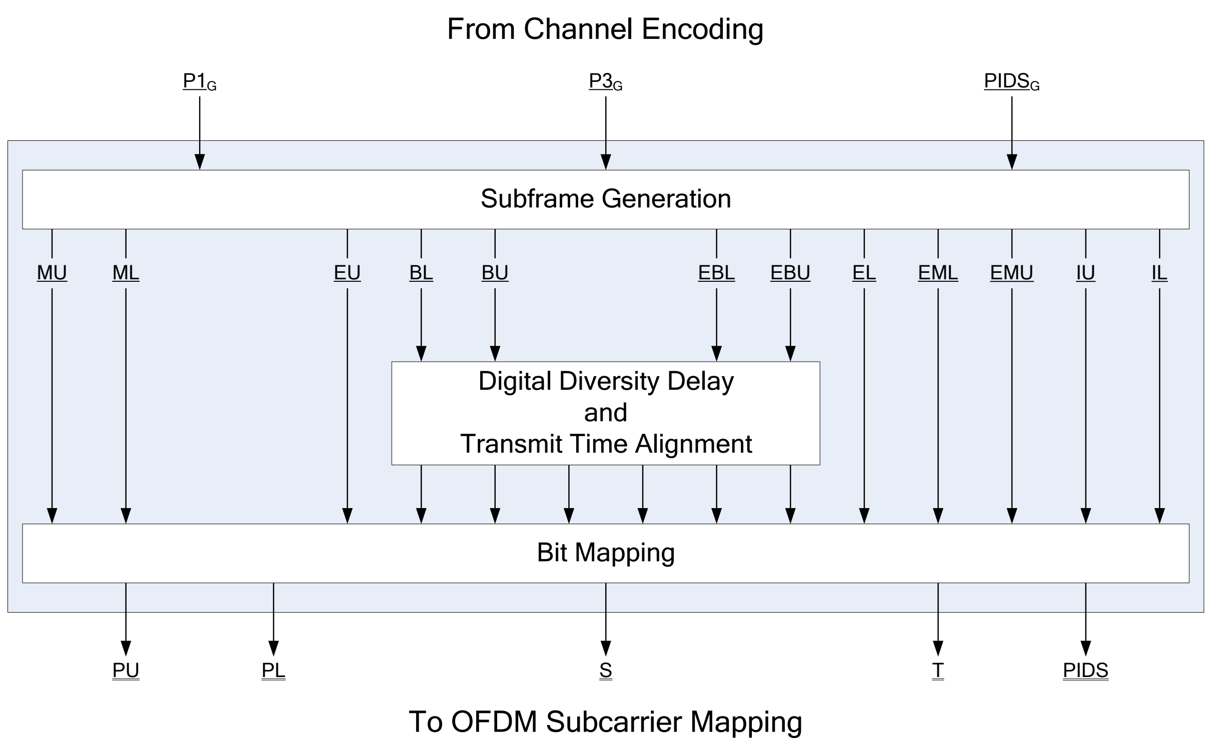

Figure 10-1: Interleaving Conceptual Block Diagram

The inputs to interleaving are the transfer frame vectors for each logical channel output from channel encoding. Table 10-1 and Table 10-2 define the size and rate of the transfer frames on each active logical channel, along with the destination interleaver matrix and the number of transfer frames required to fill the destination interleaver matrix. The output interleaver matrices are passed to OFDM Subcarrier Mapping, which maps a row from each interleaver matrix to its respective upper and lower sidebands.

Some of the interleaver matrices require more than one transfer frame to fill all of their elements. These transfer frames must be buffered until enough have been accumulated to fill the entire interleaver matrix.

In service mode MA1, eight $P1_G$ transfer frames are required to fill the PU and PL interleaver matrices. Each $P1_G$ transfer frame is delivered from Layer 2 to Layer 1 at the L1 block rate R_b. These transfer frames are buffered in subframe generation and processed at the L1 frame rate R_f. One $P3_G$ transfer frame is required to fill the S and T interleaver matrices. It is delivered from Layer 2 to Layer 1 at the L1 frame rate. To fill the PIDS interleaver matrix, $PIDS_G$ transfer frames are not buffered, but are processed at the L1 block rate.

Similar relationships for service mode MA3 can be observed in Table 10-2.

Table 10-1: Transfer Frame Characteristics – Service Mode MA1

Transfer Frame	Input Transfer Frame Size (bits)	Input Transfer Frame Rate (Hz)	Destination Interleaver Matrices	Transfer Frames per Interleaver Matrix
$P1_G$	9000	R_b	PU, PL	8
$P3_G$	36000	R_f	S, T	1
$PIDS_G$	240	R_b	PIDS	1

Table 10-2: Transfer Frame Characteristics – Service Mode MA3

Transfer Frame	Input Transfer Frame Size (bits)	Input Transfer Frame Rate (Hz)	Destination Interleaver Matrices	Transfer Frames per Interleaver Matrix
$P1_G$	9000	R_b	PU, PL	8
$P3_G$	72000	R_f	S, T	1
$PIDS_G$	240	R_b	PIDS	1

Table 10-3 summarizes the processing of the logical channels in the interleaver. Note that T_{T1a} is the transmit time alignment delay as described in Subsection 10.1.

Table 10-3: Summary of Logical Channel Processing in the Interleaver

LC	Mode	Subframe	Delay	Interleaver Depth	Interleaver Matrix
P1	MA1, MA3	BL	$T_{dd} + T_{T1a}$	Block	PL
		ML	0	Frame	
		BU	$T_{dd} + T_{T1a}$	Block	PU
		MU	0	Frame	
P3	MA1	EL	0	Frame	T
		EU	0	Frame	S
	MA3	EBL	T_{dd}	Frame	T
		EML	0	Frame	
		EBU	T_{dd}	Frame	S
		EMU	0	Frame	
PIDS	MA1, MA3	IL	0	Block	PIDS
		IU	0	Block	

10.2 Subframe Generation

The first step in the interleaving process is to split the bits of each active input transfer frame into multiple subframes and, at the same time, reorder the bits. The number of subframes for an active logical channel, the division and order of transfer frame bits, and the number of transfer frames needed to fill the subframes, are all dependent upon service mode.

The basic procedure for a given logical channel is as follows:

1. Determine the proper subframe generation structure including bit order for a given logical channel from Figure 10-4 to Figure 10-6, depending on the service mode.
2. Accumulate the indicated number of transfer frames (as shown in the figures below and indicated in Table 10-1 and Table 10-2).
3. For index = 0…N-1, where N represents the total number of input bits (equal to the input transfer frame length times the number of accumulated transfer frames), compute the indicated modulo operation and assign the indexed input bit to the proper subframe in the proper order as shown in the appropriate figure. Subframes are filled sequentially starting with index 0. Notice that the bit indexes change from the read in operation to the sub-frame filling operation.
4. Repeat Steps 1 through 3 for each active logical channel.

Depending on the service mode, some subframes may be delayed before they enter the bit mapping. See, for example, subframes <u>BL</u> and <u>BU</u> in mode MA1 (Figure 10-4).

10.3 Interleaver Matrices

A two-dimensional interleaver matrix is used to reorder and group subframe bits. The interleaving process allows individual encoded bits to be directed to specific *interleaver blocks* within the interleaver matrix. An interleaver block can be viewed as a smaller independent interleaver.

In general, an interleaver matrix has 256 rows and an interleaver block has 32 rows. Thus, there are eight interleaver blocks per interleaver matrix. The number of columns in an interleaver matrix depends on the specific matrix. The AM system uses five different interleaver matrices; PU, PL, S, T, and PIDS. Each of the matrices has 25 columns, except the PIDS matrix, which has only two. In addition, the number of bits in each element of a matrix depends not only on the interleaver matrix but also on the service mode. The number of bits per element for each matrix in each service mode is shown in Table 10-4.

Table 10-4: Number of Bits per Interleaver Matrix Element

Service Mode	Interleaver Matrix				
	PU	**PL**	**S**	**T**	**PIDS**
MA1	6	6	4	2	4
MA3	6	6	6	6	4

10.3.1 PU, PL, S, and T Interleaver Matrices

For the interleaver matrices PU, PL, S, and T, interleaving within each block is performed using the following expression for the row and column indices, where the index k points to one of 750 elements within an interleaver block:

Row(k) = [11·[(9·k) MOD 25]+16·INT(k/25)+11·INT(k/50)] MOD 32

Column(k) = (9·k) MOD 25

for k = 0,…,749

Of the total of 800 (32x25) elements in an interleaver block, the remaining 50 elements are used to transmit a known *training* pattern. The row and column for these training elements are indexed using the same equations above with k being indexed from 750 to 799.

Figure 10-2 shows a 32x25 interleaver block. The number in each element represents the value of k, which produces the row and column index for that element. “T” represents elements containing a training symbol. The bit definition of training symbols for each logical channel in each service mode is given in Table 10-5.

Column (k)

Row (k)

	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24
0	0	"T"	728	692	631	595	534	498	437	376	340	279	243	182	146	85	49	"T"	702	666	605	569	508	472	411
1	150	114	53	17	"T"	745	684	648	587	526	490	429	393	332	296	235	199	138	77	41	"T"	719	658	622	561
2	300	264	203	167	106	70	9	"T"	737	676	640	579	543	482	446	385	349	288	227	191	130	94	33	"T"	711
3	450	414	353	317	256	220	159	123	62	1	"T"	729	693	632	596	535	499	438	377	341	280	244	183	147	86
4	600	564	503	467	406	370	309	273	212	151	115	54	18	"T"	746	685	649	588	527	491	430	394	333	297	236
5	"T"	714	653	617	556	520	459	423	362	301	265	204	168	107	71	10	"T"	738	677	641	580	544	483	447	386
6	125	89	28	"T"	706	670	609	573	512	451	415	354	318	257	221	160	124	63	2	"T"	730	694	633	597	536
7	275	239	178	142	81	45	"T"	723	662	601	565	504	468	407	371	310	274	213	152	116	55	19	"T"	747	686
8	425	389	328	292	231	195	134	98	37	"T"	715	654	618	557	521	460	424	363	302	266	205	169	108	72	11
9	575	539	478	442	381	345	284	248	187	126	90	29	"T"	707	671	610	574	513	452	416	355	319	258	222	161
10	725	689	628	592	531	495	434	398	337	276	240	179	143	82	46	"T"	724	663	602	566	505	469	408	372	311
11	50	14	"T"	742	681	645	584	548	487	426	390	329	293	232	196	135	99	38	"T"	716	655	619	558	522	461
12	200	164	103	67	6	"T"	734	698	637	576	540	479	443	382	346	285	249	188	127	91	30	"T"	708	672	611
13	350	314	253	217	156	120	59	23	"T"	726	690	629	593	532	496	435	399	338	277	241	180	144	83	47	"T"
14	500	464	403	367	306	270	209	173	112	51	15	"T"	743	682	646	585	549	488	427	391	330	294	233	197	136
15	650	614	553	517	456	420	359	323	262	201	165	104	68	7	"T"	735	699	638	577	541	480	444	383	347	286
16	25	"T"	703	667	606	570	509	473	412	351	315	254	218	157	121	60	24	"T"	727	691	630	594	533	497	436
17	175	139	78	42	"T"	720	659	623	562	501	465	404	368	307	271	210	174	113	52	16	"T"	744	683	647	586
18	325	289	228	192	131	95	34	"T"	712	651	615	554	518	457	421	360	324	263	202	166	105	69	8	"T"	736
19	475	439	378	342	281	245	184	148	87	26	"T"	704	668	607	571	510	474	413	352	316	255	219	158	122	61
20	625	589	528	492	431	395	334	298	237	176	140	79	43	"T"	721	660	624	563	502	466	405	369	308	272	211
21	"T"	739	678	642	581	545	484	448	387	326	290	229	193	132	96	35	"T"	713	652	616	555	519	458	422	361
22	100	64	3	"T"	731	695	634	598	537	476	440	379	343	282	246	185	149	88	27	"T"	705	669	608	572	511
23	250	214	153	117	56	20	"T"	748	687	626	590	529	493	432	396	335	299	238	177	141	80	44	"T"	722	661
24	400	364	303	267	206	170	109	73	12	"T"	740	679	643	582	546	485	449	388	327	291	230	194	133	97	36
25	550	514	453	417	356	320	259	223	162	101	65	4	"T"	732	696	635	599	538	477	441	380	344	283	247	186
26	700	664	603	567	506	470	409	373	312	251	215	154	118	57	21	"T"	749	688	627	591	530	494	433	397	336
27	75	39	"T"	717	656	620	559	523	462	401	365	304	268	207	171	110	74	13	"T"	741	680	644	583	547	486
28	225	189	128	92	31	"T"	709	673	612	551	515	454	418	357	321	260	224	163	102	66	5	"T"	733	697	636
29	375	339	278	242	181	145	84	48	"T"	701	665	604	568	507	471	410	374	313	252	216	155	119	58	22	"T"
30	525	489	428	392	331	295	234	198	137	76	40	"T"	718	657	621	560	524	463	402	366	305	269	208	172	111
31	675	639	578	542	481	445	384	348	287	226	190	129	93	32	"T"	710	674	613	552	516	455	419	358	322	261

Figure 10-2: Interleaver Row and Column Indices vs k

Table 10-5: Training Bit Patterns

Service Mode	**Interleaver Matrix**				
	PU	**PL**	**S**	**T**	**PIDS**
MA1	100101	100101	1001	10	1001
MA3	100101	100101	100101	100101	1001

10.3.2 Interleaver Matrix PIDS

The PIDS interleaver matrix is constructed in the same way regardless of service mode. The interleaving within each interleaver block is performed using the following expressions for the row indices:

Row(k) = [11·(k+INT(k/15))+3] MOD 32

for k = 0,…,29

The index k points to one of 30 elements within an interleaver block. Of the total of 32 elements in a block, the remaining two elements are used to transmit a known training pattern. The rows for these training elements are indexed using the above equation with k being indexed from 30 to 31.

In all service modes, the subframes IU and IL are used to fill the PIDS interleaver matrix. IL fills column 0 and IU fills column 1.

From Table 10-4, each element of the PIDS interleaver matrix contains 4 bits. Elements that are reserved for training are filled with the bit pattern shown in Table 10-5. Figure 10-3 shows a 32x1 PIDS interleaver column, illustrating the structure of IU and IL. The number in each element represents the value of k, which produces the row index for that element. “T” represents elements containing a training symbol.

Row(k)	0
0	22
1	25
2	28
3	0
4	3
5	6
6	9
7	12
8	"T"
9	17
10	20
11	23
12	26
13	29
14	1
15	4
16	7
17	10
18	13
19	15
20	18
21	21
22	24
23	27
24	"T"
25	2
26	5
27	8
28	11
29	14
30	16
31	19

Figure 10-3: PIDS Interleaver Row Indices vs. k

10.4 Bit Mapping

Bit mapping accepts two or more binary subframe vectors and maps each bit in each subframe to a unique location in the destination interleaver matrix. This location includes a row, a column, and a bit position within the element determined by the row and column. This reordering of bits before transmission mitigates the impact of burst errors caused by signal fades and interference.

Figure 10-4 through Figure 10-6 show how to generate the block number b that determines the interleaver block within an interleaver matrix, the index k that determines the row and column in an interleaver block, and the index p that determines the bit position within the interleaver block element for each logical channel in each service mode. The bit mapping process is as follows:

1. For a given subframe vector in a given service mode, select the proper bit mapping equations from Figure 10-4 through Figure 10-6.
2. For n = 0…L-1, where L represents the subframe vector length, compute the parameters b, k, and p.
3. Compute the row and column indices for the destination interleaver matrix using the equations in Subsection 10.3.2 (for PIDS) or Subsection 10.3.1 (for all other matrices).
4. Transfer the subframe bit of index n to the destination interleaver matrix position at the computed row, column, block, and position indices.
5. Repeat steps (1) through (4) for all logical channels destined for a particular interleaver matrix.
6. Populate the interleaver matrix with the proper training symbols defined in Table 10-5.

10.5 Transfer Frame Time Alignment

Interleaving must maintain a specific transfer frame alignment and synchronization at its output. For a given logical channel, the BC range m1:m2 indicates which L1 blocks are spanned by the designated transfer frame. The ALFN n is the absolute L1 frame number.

10.6 Service Mode MA1

In service mode MA1, the PU, PL, S, T, interleaver matrices are populated as shown in Figure 10-4 and the PIDS interleaver matrix is populated as shown in Figure 10-5.

The PL interleaver matrix is populated with subframes BL and ML. The top set of equations from Figure 10-4 are used to determine the block index b, the index k and the bit position p for the BL subframe. Similarly, the second set of equations from the top is used for the ML subframe. The subframe index n ranges from 0 to 17,999 for both subframes. Before populating PL, subframe BL is delayed by $T_{dd} + T_{T1a}$. From Table 10-4, each element of the PL interleaver matrix contains 6 bits. Row and column indices are computed from the equations in Subsection 10.3.1. Elements that are reserved for training are filled with the bit pattern 100101 obtained from Table 10-5.

The other interleaver matrices are populated in a similar fashion as shown in Figure 10-4 and Figure 10-5.

10.7 Service Mode MA3

In service mode MA3, the PU, PL, S, T, and PIDS interleaver matrices are populated for input to the OFDM subcarrier mapping. Refer to Figure 10-6 and Figure 10-5 for details.

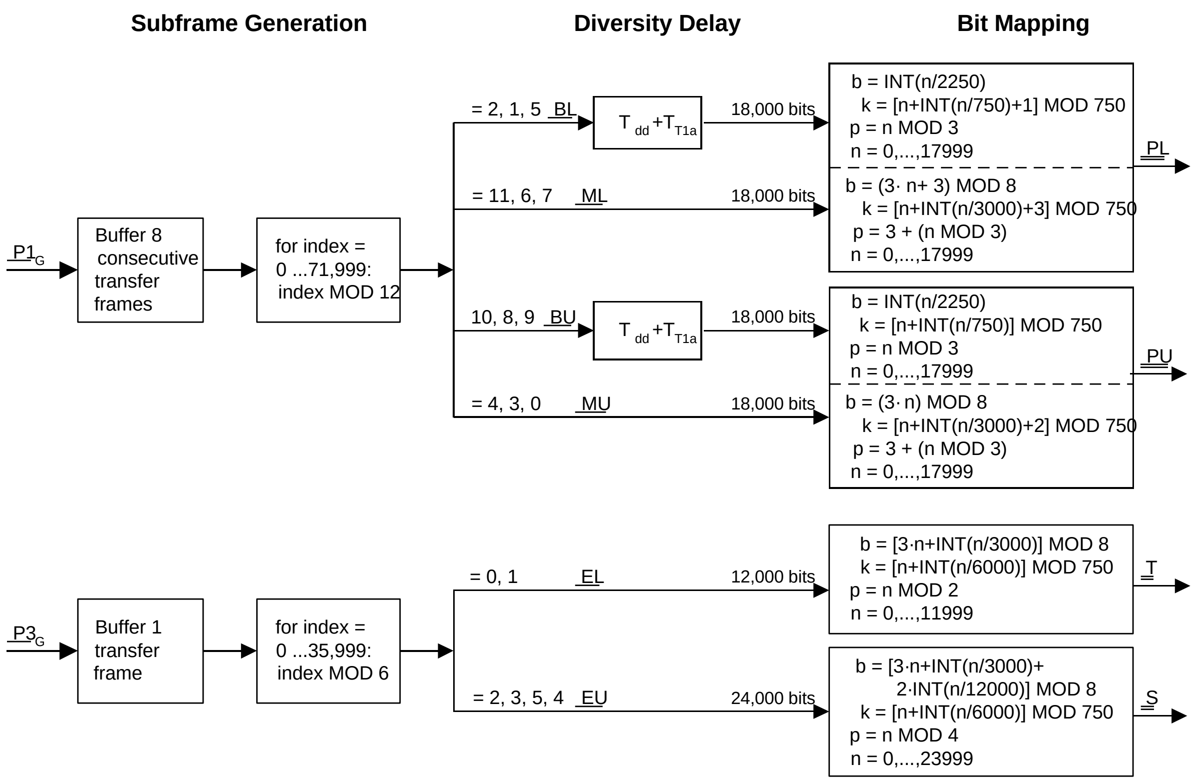

Figure 10-4: Interleaving – Service Mode MA1

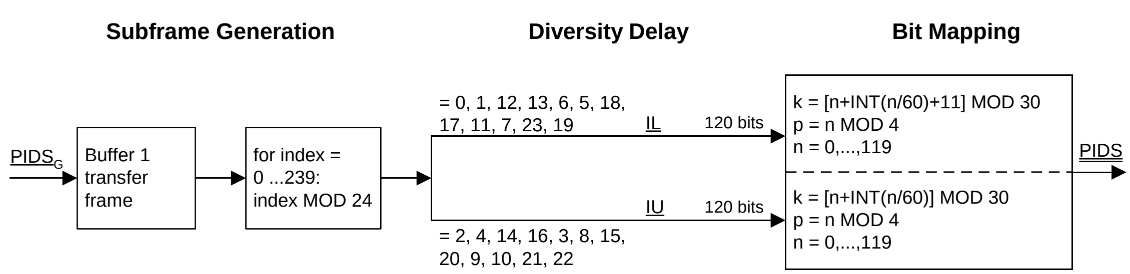

Figure 10-5: PIDS Interleaving Process – All Service Modes

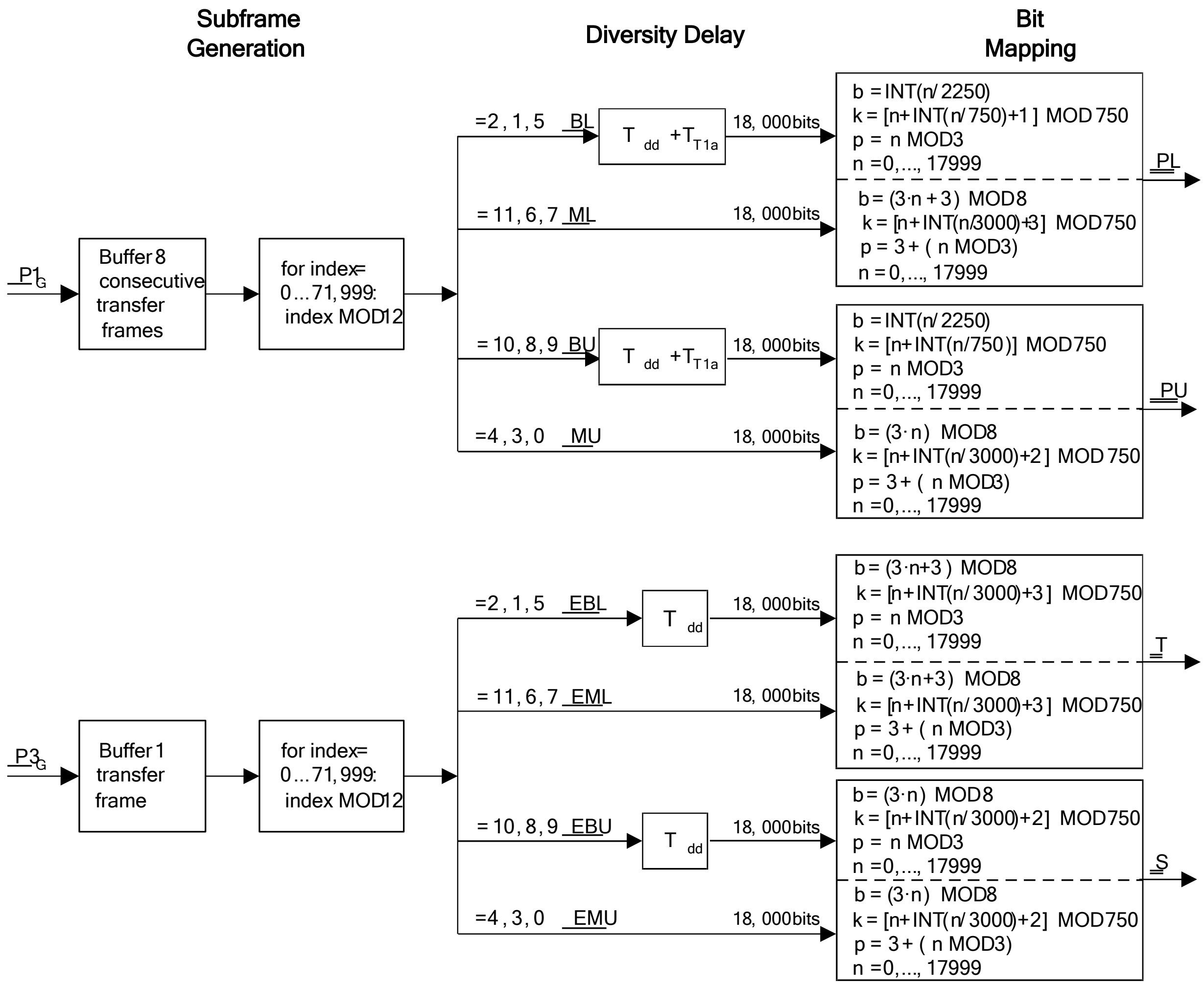

Figure 10-6: Interleaving – Service Mode MA3

11 System Control Processing

11.1 Introduction

System control processing receives system control data from the Configuration Administrator via the SCCH. This data is combined with synchronization, *parity*, and reserved bits within Layer 1 to create system control data sequences. The resulting sequences are destined for the reference subcarriers located on both sides of the main analog carrier.

The resulting output is a column vector $\underline{R}$ of fixed dimension. The number of elements of $\underline{R}$ corresponds to the number of OFDM symbols per L1 frame. The vector $\underline{R}$ is comprised of eight 32-bit sequences, one for each L1 block, and is output at the L1 frame rate, R_f. In addition, system control processing provides, via the SCCH, the L1 block count to Layer 2 at the L1 block rate, R_b. A conceptual view of the system control processing is shown in Figure 11-1.

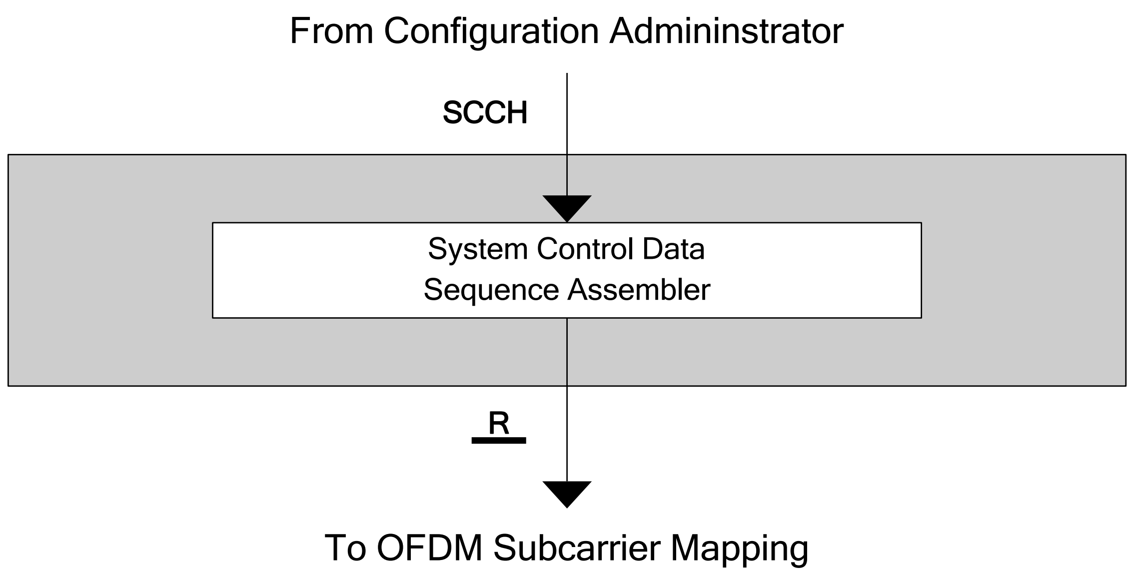

Figure 11-1: System Control Processing Conceptual Diagram

11.2 System Control Data Sequence Assembler

The system control data sequence assembler collects information from the Configuration Administrator and creates a vector of eight 32-bit system control data sequences. Each element of the vector contains one bit. Each of the eight data sequences is comprised of bit fields that represent various system control components as well as synchronization and timing information. The system control data sequence is depicted in Figure 11-2 and defined in Table 11-1.

The subsections that follow this section describe each of the bit fields. Bits 31 to 0 of the system control data sequence, map to bits 0 to 31, 32 to 63, 64 to 95, 96 to 127, 128 to 159, 160 to 191, 192 to 223, and 224 to 255 of $\underline{R}$, respectively. Refer to Subsection 12.2.3 for a discussion of how $\underline{R}$ is mapped to the reference subcarriers.

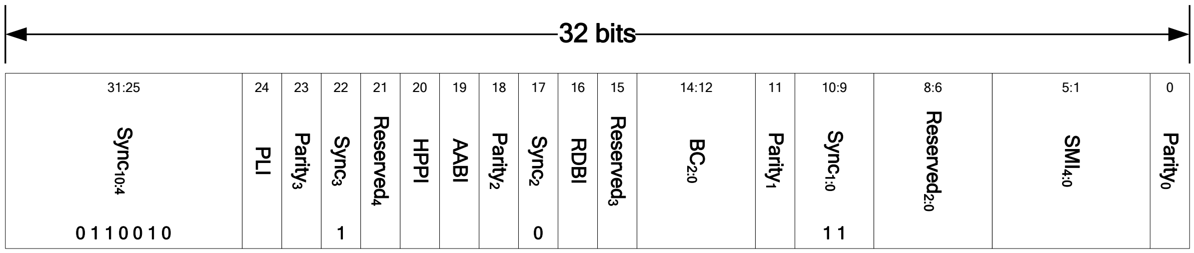

Figure 11-2: System Control Data Sequence

Table 11-1: System Control Data Sequence Bit Allocations

Field	Bit Location	Bit Length	Description
Sync$_{10:4}$	31:25	7	Sync$_{10:4}$ = 0110010
Power Level Indicator (PLI)	24	1	0 = Hybrid carriers scaled by CH_{S1}, CH_{I1}, and $\underline{CH}_{T1}$ 1 = Hybrid carriers scaled by CH_{S2} CH_{I2}, and $\underline{CH}_{T2}$
Parity$_3$	23	1	Even Parity for PLI
Sync$_3$	22	1	Sync$_3$ = 1
Reserved$_4$	21	1	For future expansion
HPPI	20	1	High-Power PIDS Indicator
Analog Audio Bandwidth Indicator (AABI)	19	1	0 = 5 kHz analog audio bandwidth 1 = 8 kHz analog audio bandwidth
Parity$_2$	18	1	Even Parity for Reserved$_4$, HPPI, and AABI
Sync$_2$	17	1	Sync$_2$ = 0
RDBI	16	1	Reduced Digital Bandwidth Indicator
Reserved$_3$	15	1	For future expansion
Block Count (BC$_{2:0}$)	14:12	3	Modulo-8 count which increments every 32 OFDM symbols
Parity$_1$	11	1	Even Parity for RDBI, Reserved$_3$ and Block Count
Sync$_{1:0}$	10:9	2	Sync$_{1:0}$ = 11
Reserved$_{2:0}$	8:6	3	For future expansion
Service Mode Indicator (SMI$_{4:0}$)	5:1	5	Identifies service mode currently selected
Parity$_0$	0	1	Even parity for Reserved$_{2:0}$ and SMI

11.2.1 Block Synchronization (Sync)

The block synchronization (sync) bits serve to aid in receiver synchronization. The sync bit pattern is distributed over the system control data sequence as shown in Table 11-1 and Figure 11-2.

11.2.2 Power Level Indicator (PLI)

The power level indicator (PLI) is a one-bit flag used to indicate the nominal level of the secondary and tertiary sidebands when transmitting a Hybrid waveform. If the flag is cleared, the subcarriers in the secondary, PIDS, and tertiary sideband have been scaled by CH_{S1}, CH_{I1}, and $\underline{CH}_{T1}$; if the flag is set, these subcarriers have been scaled by CH_{S2}, CH_{I2}, and $\underline{CH}_{T2}$. See Reference [9] for detailed information concerning the levels of the sidebands. The PLI flag is always 0 when an All Digital waveform is being transmitted.

When RDBI is set to 1, PLI is not applicable and shall always be zero. Otherwise, PLI reflects the current configuration as controlled by the PL control signal received from the configuration administrator. Refer to Subsection 6.3 for specific details on how PL affects the spectral profile.

11.2.3 High-Power PIDS Indicator (HPPI)

The High Power PIDS Indicator (HPPI) is a one-bit flag used to indicate the level of the PIDS2/PIDS2* subcarriers in service mode MA1 and the PIDS1/PIDS2 subcarriers in service mode MA3. If HPPI is 0, these PIDS subcarriers are in the standard low-power configuration. If HPPI is 1, these PIDS subcarriers are in the high-power configuration.

When RDBI is set to 1, HPPI is not applicable and shall always be 0. Otherwise, HPPI reflects the current configuration as controlled by the HPP control signal received from the configuration administrator. Refer to Subsection 6.3 for specific details on how HPP affects the spectral profile.

11.2.4 Analog Audio Bandwidth Indicator (AABI)

The analog audio bandwidth indicator (AABI) is a one-bit flag used to indicate the maximum bandwidth of the analog audio signal when transmitting a Hybrid waveform. If the flag is cleared, the maximum analog audio bandwidth is 5 kHz; if the flag is set, the maximum analog audio bandwidth is 8 kHz. The AABI flag is always 0 when an All Digital waveform is being transmitted.

11.2.5 Reduced Digital Bandwidth Indicator (RDBI)

The Reduced Digital Bandwidth Indicator (RDBI) is a one-bit flag to indicate the spectral profile of the MA1 and MA3 waveform. When RDBI is 0 (standard configuration), the secondary and tertiary sidebands are enabled, and the PIDS subcarriers are all active.

When RDBI is 1 (reduced bandwidth configuration), the secondary and tertiary sidebands are disabled. In addition, for service mode MA1, the PIDS1 and PIDS1* subcarriers are disabled. In both service modes MA1 and MA3, when RDBI is 1, the active PIDs subcarriers are in the high-power configuration.

When RDBI is set to 1, the state of the PLI and HPPI indicators shall be zero.

RDBI reflects the current configuration as controlled by the RDB control signal received from the configuration administrator. Refer to Subsection 6.3 for specific details on how RDB affects the spectral profile.

11.2.6 L1 Block Count (BC)

The L1 Block Count ($BC_{2:0}$) is a modulo-8 counter indicating the current L1 block within an L1 frame. The L1 Block Count increments on each 32-OFDM-symbol boundary. An L1 Block Count of 0 signifies the start of an L1 frame, while an L1 Block Count of 7 signifies the final block of an L1 frame. The first L1 Block Count inserted into the system control data sequence is 0. The L1 Block Count bit assignments are shown in Table 11-2.

Table 11-2: L1 Block Count Bit Assignments

L1 Block Count	$BC_{2:0}$ Bit Assignment (Bits 14:12)
0	000
1	001
2	010
3	011
4	100
5	101
6	110
7	111

11.2.7 Service Mode Indicator (SMI)

The AM HD Radio system supports one Hybrid service mode (MA1) and one All Digital service mode (MA3), as defined in Section 6. The service mode indicator ($SMI_{4:0}$) is a five-bit field that uniquely indicates the current service mode. The definition of $SMI_{4:0}$ for each service mode is shown in Table 11-3. Values 00011 though 11111 are reserved.

Table 11-3: AM HD Radio Service Mode Bit Assignments

Service Mode	Service Mode Indicator (Bits 4:0)
None	00000
MA1	00001
MA3	00010
Reserved	00011 - 11111

Each of the reserved service modes must maintain backward compatibility as defined in Subsection 6.2.1. Thus first generation receivers will always configure themselves to one of the basic operational modes MA1 or MA3 when one of the reserved modes is detected. The one exception to this is when the two least significant bits of PSMI are both zero. In this case, backward compatibility is not maintained. This case is reserved for system test modes and operational receivers will assume that the service mode is “None” and all outputs will be disabled.

12 OFDM Subcarrier Mapping

12.1 Introduction

OFDM subcarrier mapping transforms rows of interleaver matrices into scaled *QPSK*, 16-QAM or 64-QAM symbols in service modes MA1 and MA3 and maps them to specific OFDM subcarriers in the output vector X. System control data sequence bits from R are transformed into *BPSK* symbols and mapped to the two reference subcarrier locations in X. Figure 12-1 shows the inputs, output, and component functions of OFDM subcarrier mapping.

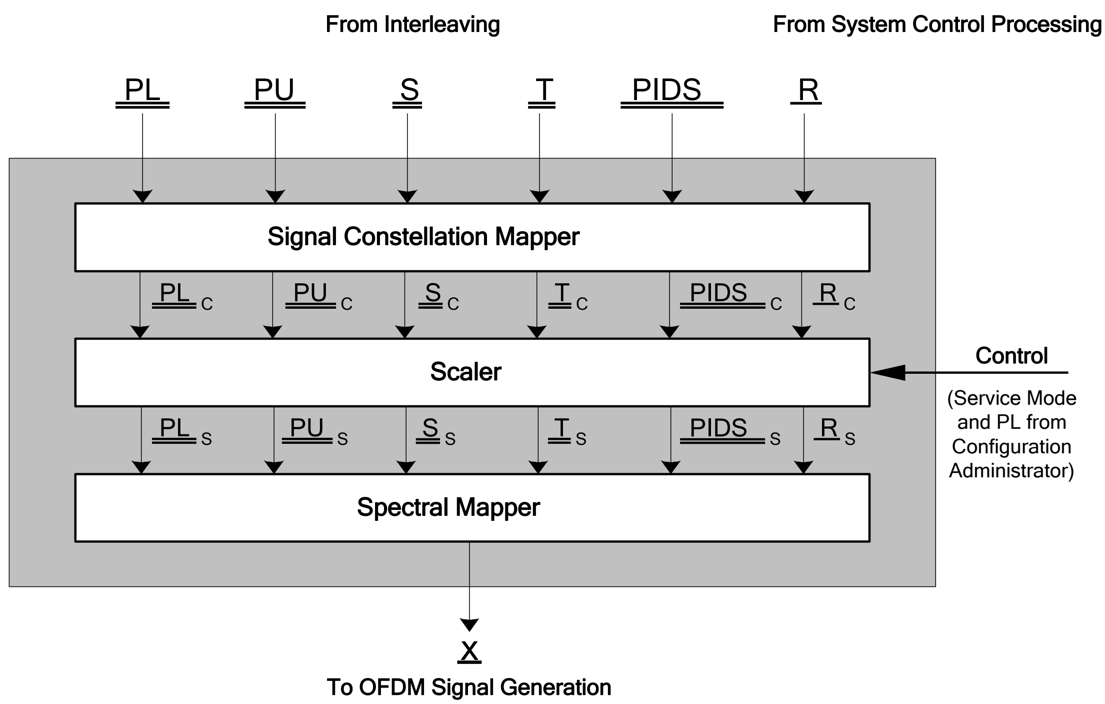

Figure 12-1: OFDM Subcarrier Mapping Functional Block Diagram

The inputs to OFDM subcarrier mapping are the interleaver matrices PL, PU, S, T, PIDS, and R, the system control data sequence vector. Matrices S, T, PIDS, and R are mapped to the secondary, tertiary, IDS, and reference subcarriers respectively. Matrices PL and PU are mapped to the primary lower and primary upper subcarriers, respectively.

The output from OFDM subcarrier mapping for each OFDM symbol is a single complex vector, X, of length 163. The vector is indexed by $k = 0,1,2,...,162$. The kth element of X corresponds to subcarrier $(k - 81)$, as shown in Figure 12-2.

Index into X	0	1	2	. . .	160	161	162
Subcarrier Number	-81	-80	-79		79	80	81

Figure 12-2: Assignment of Output Vector X Elements to Subcarriers

The elements of X are populated based on service mode. Elements corresponding to unused subcarriers are set to the complex value 0 + j0. Elements of R and the associated row from each active interleaver matrix are assigned to the same instance of X.

The first block in Figure 12-1 is the *signal constellation mapper*, which converts the individual elements of the various interleaver matrices from digital words to complex values representing constellation points. The dimensions of the interleaver matrices, therefore, remain unchanged during this process. However, to distinguish between the interleaver matrices before and after their elements have been transformed to constellation values, a subscript "C" is added after the mapping has been performed.

The next block in Figure 12-1 is the scaler. This block multiplies each complex element of the interleaver matrices by a scale factor to place the subcarriers at the appropriate level relative to the unmodulated carrier. To indicate that scaling has been applied, a subscript "S" is added to the interleaver matrices.

The final block in Figure 12-1 is the spectral mapper. This block places the scaled constellation values in the appropriate position in the output vector $\underline{X}$.

12.2 OFDM Subcarrier Mapping Procedures

The details of how the elements of each interleaver matrix get mapped to complex constellation values is described in Subsections 12.2.1, 12.2.2, and 12.2.3, beginning with the primary interleaver matrices.

For each active interleaver matrix, a row is processed every T_s seconds. Rows are processed sequentially, starting with the first row (Row 0). When all rows of an interleaver matrix have been processed, the next instance of that interleaver matrix is processed, starting with the first row. The columns of an interleaver matrix map directly to OFDM subcarriers. Subcarriers -56, -55, -54, 0, 54, 55, and 56 are never used and therefore are always set to (0 + j0).

12.2.1 Primary Subcarriers

The 256x25 primary interleaver matrices ($\underline{\underline{PU}}$ and $\underline{\underline{PL}}$) are mapped to 64-QAM constellations. The elements of these matrices are six-bit words. To map each six-bit word within an interleaver matrix to a subcarrier location within the vector $\underline{X}$, the following steps are taken:

1. Read a six-bit word from an element within an interleaver matrix.
2. Map the six-bit word from Step 1 to a complex constellation value using Table 12-1.
3. Scale the I and Q components of the complex constellation value from Step 2 using the appropriate amplitude scale factor from Table 12-11.
4. Map the scaled constellation value from Step 3 to the appropriate element of X using Table 12-2 or Table 12-3.

There is a one-to-one mapping from primary interleaver matrix columns to OFDM subcarriers. In both the Hybrid and All Digital waveforms, $\underline{\underline{PU}}_S$ and $\underline{\underline{PL}}_S$ map to the primary sidebands as shown in Table 12-2 and Table 12-3. In each case, the mapping begins with the subcarrier index with the lowest absolute value and continues to the subcarrier index with the highest absolute value. In the Hybrid waveform, the first element of $\underline{\underline{PU}}_S$ maps to subcarrier 57 and the first element of $\underline{\underline{PL}}_S$ maps to subcarrier -57. In the All Digital waveform, the first element of $\underline{\underline{PU}}_S$ maps to subcarrier 2 and the first element of $\underline{\underline{PL}}_S$ maps to subcarrier -2. In addition, the constellation values of the lower sidebands are negated and complex conjugated. This is indicated in Table 12-2 and Table 12-3 by a minus sign and an asterisk appended to the interleaver matrix designation, for example, $-\underline{\underline{PL}}_S$*.

Table 12-1: 64-QAM Constellation Mapping

6 bit word $x_5x_4x_3x_2x_1x_0$	**Hex**	**Constellation Value**	**6 bit word** $x_5x_4x_3x_2x_1x_0$	**Hex**	**Constellation Value**
000000	0	-3.5 - j3.5	100000	20	-3.5 - j2.5

6 bit word $x_5x_4x_3x_2x_1x_0$	Hex	Constellation Value	6 bit word $x_5x_4x_3x_2x_1x_0$	Hex	Constellation Value
000001	1	3.5 - j3.5	100001	21	3.5 - j2.5
000010	2	-0.5 - j3.5	100010	22	-0.5 - j2.5
000011	3	0.5 - j3.5	100011	23	0.5 - j2.5
000100	4	-2.5 - j3.5	100100	24	-2.5 - j2.5
000101	5	2.5 - j3.5	100101	25	2.5 - j2.5
000110	6	-1.5 - j3.5	100110	26	-1.5 - j2.5
000111	7	1.5 - j3.5	100111	27	1.5 - j2.5
001000	8	-3.5 + j3.5	101000	28	-3.5 + j2.5
001001	9	3.5 + j3.5	101001	29	3.5 + j2.5
001010	A	-0.5 + j3.5	101010	2A	-0.5 + j2.5
001011	B	0.5 + j3.5	101011	2B	0.5 + j2.5
001100	C	-2.5 + j3.5	101100	2C	-2.5 + j2.5
001101	D	2.5 + j3.5	101101	2D	2.5 + j2.5
001110	E	-1.5 + j3.5	101110	2E	-1.5 + j2.5
001111	F	1.5 + j3.5	101111	2F	1.5 + j2.5
010000	10	-3.5 - j0.5	110000	30	-3.5 - j1.5
010001	11	3.5 - j0.5	110001	31	3.5 - j1.5
010010	12	-0.5 - j0.5	110010	32	-0.5 - j1.5
010011	13	0.5 - j0.5	110011	33	0.5 - j1.5
010100	14	-2.5 - j0.5	110100	34	-2.5 - j1.5
010101	15	2.5 - j0.5	110101	35	2.5 - j1.5
010110	16	-1.5 - j0.5	110110	36	-1.5 - j1.5
010111	17	1.5 - j0.5	110111	37	1.5 - j1.5
011000	18	-3.5 + j0.5	111000	38	-3.5 + j1.5
011001	19	3.5 + j0.5	111001	39	3.5 + j1.5
011010	1A	-0.5 + j0.5	111010	3A	-0.5 + j1.5
011011	1B	0.5 + j0.5	111011	3B	0.5 + j1.5
011100	1C	-2.5 + j0.5	111100	3C	-2.5 + j1.5
011101	1D	2.5 + j0.5	111101	3D	2.5 + j1.5
011110	1E	-1.5 + j0.5	111110	3E	-1.5 + j1.5
011111	1F	1.5 + j0.5	111111	3F	1.5 + j1.5

Table 12-2: Primary Interleaver Subcarrier Mapping – Hybrid Waveform

Mode	Starting Subcarrier Number	Ending Subcarrier Number	Interleaver Matrix	Interleaver Matrix Starting Column Number	Interleaver Matrix Ending Column Number
MA1	-57	-81	$-\underline{\underline{PL}}_S{}^*$	0	24
MA1	57	81	$\underline{\underline{PU}}_S$	0	24

Table 12-3: Primary Interleaver Subcarrier Mapping – All Digital Waveform

Mode	Starting Subcarrier Number	Ending Subcarrier Number	Interleaver Matrix	Interleaver Matrix Starting Column Number	Interleaver Matrix Ending Column Number
MA3	-2	-26	$-\underline{\underline{PL}}_S{}^*$	0	24
MA3	2	26	$\underline{\underline{PU}}_S$	0	24

12.2.2 Secondary and Tertiary Subcarriers

In both the Hybrid and All Digital waveforms, $\underline{\underline{S}}$ maps to the secondary sideband and $\underline{\underline{T}}$ maps to the tertiary sideband. However, in the Hybrid waveform there is not a one-to-one mapping and different modulation types are employed. For these reasons the Hybrid and All Digital waveforms are treated in separate subsections.

12.2.2.1 Hybrid Waveform

The secondary and $\underline{\underline{PIDS}}$ interleaver matrices are mapped to 16-QAM constellations while the tertiary interleaver matrix is mapped to QPSK constellations. $\underline{\underline{S}}$ and $\underline{\underline{T}}$ have dimensions of 256x25 and each element of $\underline{\underline{S}}$ contains a four-bit word while the elements of $\underline{\underline{T}}$ contain two-bit words. The $\underline{\underline{PIDS}}_S$ matrix has dimensions of 32x2 and consists of four-bit words.

To map each four-bit, or two-bit, word within an interleaver matrix to a subcarrier location within the vector $\underline{X}$, the following steps are taken:

1. Read a four-bit, or two-bit, word from a column within an interleaver matrix.
2. Map the four-bit, or two-bit, word from Step 1 to a complex constellation value using Table 12-4 for two-bit words and Table 12-5 for four-bit words.
3. Scale the I and Q components of the complex constellation value from Step 2 using the appropriate amplitude scale factor from Table 12-11. Where more than one scale factor is indicated, selection is determined by PL, input from the Configuration Administrator via the SCCH.
4. Map the scaled constellation value from Step 3 to the appropriate element of $\underline{X}$ using Table 12-6 or Table 12-7.

The procedure for mapping the constellation values in the elements of $\underline{\underline{S}}_S$ and $\underline{\underline{T}}_S$ to the secondary and tertiary sidebands for the Hybrid waveform is as follows. First, map the elements of $\underline{\underline{S}}_S$ onto the secondary upper sideband starting with the lowest subcarrier index and continuing until all columns of $\underline{\underline{S}}_S$ in the row of interest are mapped. Next, map the negated complex conjugated elements of $\underline{\underline{S}}_S$ onto the lower secondary sideband starting with the subcarrier index with the lowest absolute value and continuing until all columns of $\underline{\underline{S}}_S$ in the row of interest are mapped. Repeat this procedure for $\underline{\underline{T}}_S$, and the tertiary sidebands.

The $\underline{\underline{PIDS}}_S$ matrix also maps to the secondary sidebands. The first interleaver column (0) of $\underline{\underline{PIDS}}_S$ maps to the carriers ±27 and the second column maps to carriers ±53. Each column maps to two carriers. Therefore, the constellations on the negative carriers are first negated and complex conjugated as shown in Table 12-7.

Table 12-4: QPSK Constellation Mapping

2 bit word x_1x_0	**Hex**	**Constellation Value**
00	0	-0.5 - j0.5
01	1	0.5 - j0.5
10	2	-0.5 + j0.5
11	3	0.5 + j0.5

Table 12-5: 16-QAM Constellation Mapping

4 bit word $x_3x_2x_1x_0$	**Hex**	**Constellation Value**
0000	0	-1.5 - j1.5
0001	1	1.5 - j1.5
0010	2	-0.5 - j1.5
0011	3	0.5 - j1.5
0100	4	-1.5 + j1.5
0101	5	1.5 + j1.5
0110	6	-0.5 +j1.5
0111	7	0.5 + j1.5
1000	8	-1.5 - j0.5
1001	9	1.5 - j0.5
1010	A	-0.5 - j0.5
1011	B	0.5 - j0.5
1100	C	-1.5 + j0.5
1101	D	1.5 + j0.5
1110	E	-0.5 + j0.5
1111	F	0.5 + j0.5

Table 12-6: Secondary and Tertiary Interleaver Subcarrier Mapping – Hybrid Waveform

Starting Subcarrier Number	Ending Subcarrier Number	Interleaver Matrix	Interleaver Matrix Starting Column Number	Interleaver Matrix Ending Column Number
2	26	$\underline{\underline{T}}_S$	0	24
28	52	$\underline{\underline{S}}_S$	0	24
-2	-26	$-\underline{\underline{T}}_S^*$	0	24
-28	-52	$-\underline{\underline{S}}_S^*$	0	24

Table 12-7: PIDS Interleaver Subcarrier Mapping – Hybrid Waveform

Subcarrier Number	Interleaver Matrix	Interleaver Matrix Column Number
-27	$-\underline{\underline{PIDS}}_S^*$	0
-53	$-\underline{\underline{PIDS}}_S^*$	1
27	$\underline{\underline{PIDS}}_S$	0
53	$\underline{\underline{PIDS}}_S$	1

12.2.2.2 All Digital Waveform

In the All Digital waveform, the secondary and tertiary interleaver matrices are mapped to 64-QAM constellations while the $\underline{\underline{PIDS}}$ interleaver matrix is mapped to 16-QAM constellations. $\underline{\underline{S}}$ and $\underline{\underline{T}}$ have dimensions of 256x25 and each element contains a six-bit word. The $\underline{\underline{PIDS}}$ matrix has elements consisting of four-bit words and has dimensions 32x2.

To map each six/four-bit word within an interleaver matrix to a subcarrier location within the vector X, the following steps are taken:

1. Read a six/four-bit word from a column within an interleaver matrix.
2. Map the six/four-bit word from Step 1 to a complex constellation value using Table 12-1 for six-bit words and Table 12-5 for four-bit words.
3. Scale the I and Q components of the complex constellation value from Step 2 using the appropriate amplitude scale factor from Table 12-11.
4. Map the scaled constellation values from Step 3 to the appropriate element of X using Table 12-8 or Table 12-9.

There is a one-to-one mapping from secondary, tertiary, and $\underline{\underline{PIDS}}$ interleaver matrix columns to OFDM subcarriers. The secondary interleaver maps to the secondary sideband, the tertiary interleaver matrix maps to the tertiary sideband, and the PIDS interleaver matrix maps to both secondary and tertiary sidebands. For the $\underline{\underline{PIDS}}$ interleaver matrix, the first column maps to the index with the lowest absolute value in the tertiary (-27) sideband and the second column maps to the subcarrier index with the lowest absolute value in the secondary sideband (+27) as indicated in Table 12-9. The mapping for both $\underline{\underline{S}}_S$ and $\underline{\underline{T}}_S$ begins with the subcarrier index with the second lowest absolute value and continues to the subcarrier with the highest absolute value. As with the primary lower interleaver matrix, the constellation of $\underline{\underline{T}}_S$ is negated and complex conjugated. Similarly the column of the PIDS matrix that gets mapped to -27 is also negated and complex conjugated. The mapping of $\underline{\underline{S}}_S$ and $\underline{\underline{T}}_S$ for the All Digital waveform is summarized in Table 12-8.

Table 12-8: Secondary and Tertiary Interleaver Subcarrier Mapping – All Digital Waveform

Starting Subcarrier Number	Ending Subcarrier Number	Interleaver Matrix	Interleaver Matrix Starting Column Number	Interleaver Matrix Ending Column Number
-28	-52	$-\underline{\underline{T}}_S^*$	0	24
28	52	$\underline{\underline{S}}_S$	0	24

Table 12-9: $\underline{\underline{PIDS}}$ Interleaver Subcarrier Mapping – All Digital Waveform

Subcarrier Number	Interleaver Matrix	Interleaver Matrix Column Number
-27	$-\underline{\underline{PIDS}}_S^*$	0
27	$\underline{\underline{PIDS}}_S$	1

12.2.3 Reference Subcarriers

The input vector $\underline{R}$ consists of 256 bits (that is, a single bit for every OFDM symbol in an L1 frame). The bits of this vector are mapped to BPSK constellation points as shown in Table 12-10.

To map each bit within the $\underline{R}$ vector to a subcarrier location within the vector $\underline{X}$, the following steps are taken:

1. Read a bit from the vector $\underline{R}$.
2. Map the bit from Step 1 to a complex constellation value using Table 12-10.
3. Scale the I and Q components of the complex constellation value from Step 2 using the appropriate amplitude scale factor from Table 12-11.
4. Map the scaled constellation value from Step 3 to the appropriate elements of X using Table 12-12.

Table 12-10: BPSK Signal Constellation Mapping

n^{th} Bit Value	n^{th} Constellation Point
0	0 - j0.5
1	0 + j0.5

Table 12-11: Signal Constellation Scale Factors

Interleaver Matrix	**Waveform**	
	Hybrid	**All Digital**
$\underline{\underline{PU}}_C$	CH_P	CD_P
$\underline{\underline{PL}}_C$	CH_P	CD_P
$\underline{\underline{S}}_C$	CH_{S1} or CH_{S2}	CD_E
$\underline{\underline{T}}_C$	$\underline{CH}_{T1}[25]$ or $\underline{CH}_{T2}[25]$	CD_E
$\underline{\underline{PIDS}}_C$	CH_{I1} or CH_{I2}	CD_I
$\underline{\underline{R}}_C$	CH_B	CD_B

Table 12-12: R Subcarrier Mapping – Hybrid and All Digital Waveforms

Subcarrier Number	**System Control Data Sequence Vector**	**Interleaver Vector Column Number**
-1	$\underline{-R}_S$*	0
1	$\underline{R}_S$	0

13 OFDM Signal Generation

13.1 Introduction

OFDM signal generation receives complex frequency-domain OFDM symbols from the output of OFDM subcarrier mapping and outputs time-domain pulses representing the digital portion of the AM HD Radio signal. A conceptual block diagram of OFDM signal generation is shown in Figure 13-1.

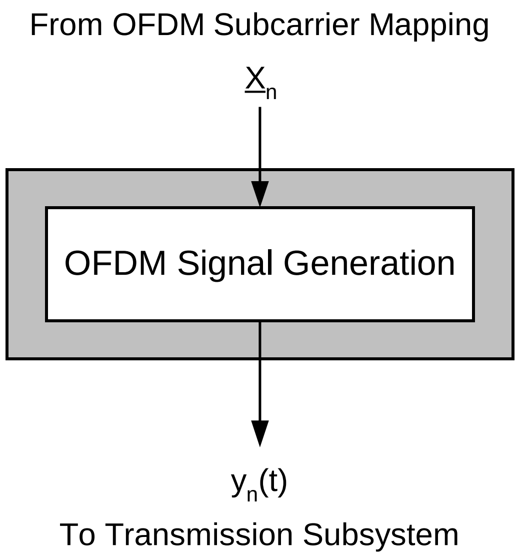

Figure 13-1: OFDM Signal Generation Conceptual Block Diagram

The input to OFDM signal generation is a complex vector, $\underline{X}_n$ of length L, representing the complex constellation values for each OFDM subcarrier in OFDM symbol n. For notational convenience, the output of OFDM Subcarrier Mapping described in Section 12 did not use the subscript n. Rather, it represented the vector $\underline{X}$ as a single OFDM symbol. In this section, the subscript is appended to $\underline{X}$ because of the significance of n to OFDM signal generation.

The output of OFDM signal generation is a complex, baseband, time-domain pulse $y_n(t)$, representing the digital portion of the AM HD Radio signal for symbol n.

13.2 Functionality

Let $\underline{X}_n[k]$ be the complex scaled constellation points from OFDM subcarrier mapping for the n^{th} symbol, where $k = 0, 1, \ldots, L$-1 indexes the OFDM subcarriers. Let $y_n(t)$ denote the complex time-domain output of OFDM signal generation for the n^{th} symbol. Then $y_n(t)$ can be written in terms of $\underline{X}_n[k]$ as follows:

$$y_n(t) = W(t - nT_s) \cdot \sum_{k=0}^{L-1} \underline{X}_n[k] \cdot e^{j2\pi\Delta f\left[k-\left(\frac{L-1}{2}\right)\right]\cdot(t-nT_S)}$$

where $n = 0,1,\ldots,\infty$, $0 \le t \le \infty$, L = 163 is the maximum number of OFDM subcarriers, and T_S and Δf are the OFDM symbol period and OFDM subcarrier spacing, respectively, as defined in Section 3.5.

The *pulse-shaping function* $W(\xi)$ is defined as:

$$W(\xi)=\begin{cases}\sqrt{\int_{-T_S}^{T_S} H(\tau)\cdot G(\xi-\tau)\cdot d\tau} & ; \quad -\frac{174}{270}\cdot T_S \le \xi \le \frac{174}{270}\cdot T_S \\ 0 & ; \quad \text{otherwise}\end{cases}$$

where

$$H(\xi)=\begin{cases}1 & ; \quad |\xi| \le \frac{1-\alpha}{2}\cdot T \\ \frac{1}{2}\cdot\left[1+\cos\left(\frac{\pi}{2\cdot\alpha}\cdot\left[2\cdot\frac{|\xi|}{T}+\alpha-1\right]\right)\right] & ; \quad \frac{1-\alpha}{2}\cdot T \le |\xi| \le \frac{1+\alpha}{2}\cdot T \\ 0 & ; \quad \text{otherwise}\end{cases}$$

and

$$G(\xi)=\frac{90}{T_S\cdot\sqrt{2\cdot\pi}}\cdot e^{-4050\cdot\left(\xi/T_S\right)^2}$$

α is the cyclic prefix width defined in Subsection 3.5, and $T = 1/\Delta f$ is the reciprocal of the OFDM subcarrier spacing. Figure 13-2 shows a plot of the pulse shaping function $W(\xi)$. Notice that the OFDM pulse shape is centered at time zero, corresponding with the center of the complex time-domain pulse $y_n(t)$. This symbol may be sampled and time-shifted for convenience such that the sample indices are positive integers. Also notice that the OFDM symbol is longer than the symbol period T_S having tails that overlap the adjacent symbols.

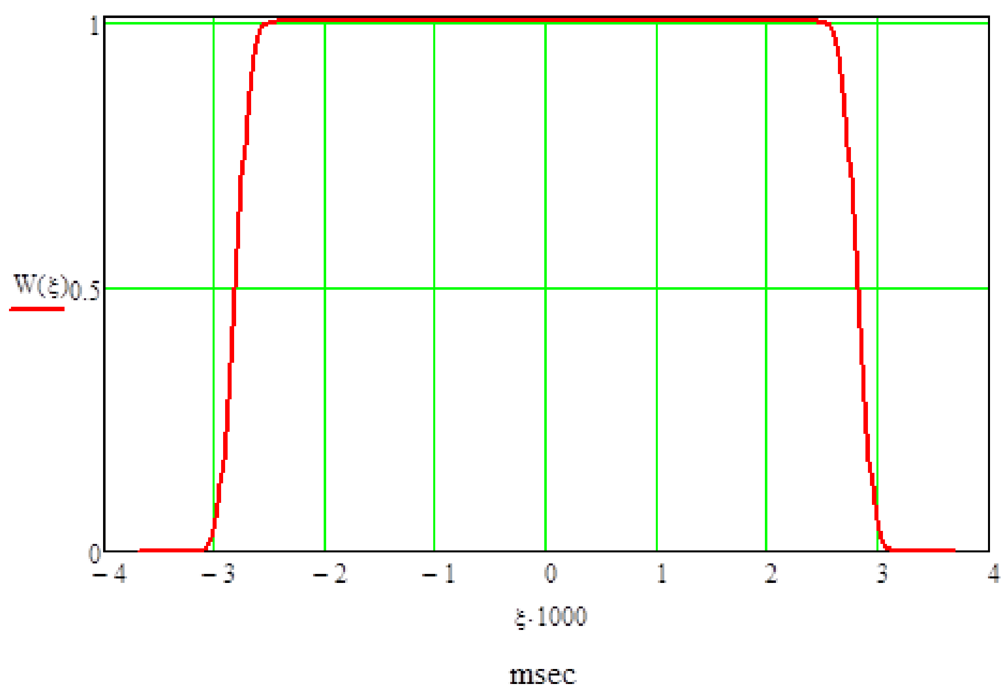

Figure 13-2: Pulse Shaping Function

14 Transmission Subsystem

14.1 Introduction

The transmission subsystem formats the baseband AM HD Radio waveform for transmission through the MF channel. Functions include symbol concatenation, and frequency up-conversion. In addition, when transmitting the Hybrid waveform, this function filters and modulates the baseband analog audio signal before coherently combining it with the digital portion of the waveform.

The input to this module is a complex, baseband, time-domain OFDM symbol, $y_n(t)$, from OFDM signal generation. A baseband analog audio signal, m(t), after application of analog diversity delay T_{ad} plus an appropriate implementation-specific *Transmit Audio Alignment Delay*, T_{T5a}, is also input from an analog source when transmitting the Hybrid waveform. The output of this module is the MF AM HD Radio waveform.

Refer to Figure 14-1 and Figure 14-2 for functional block diagrams of the Hybrid and All Digital transmission subsystems, respectively.

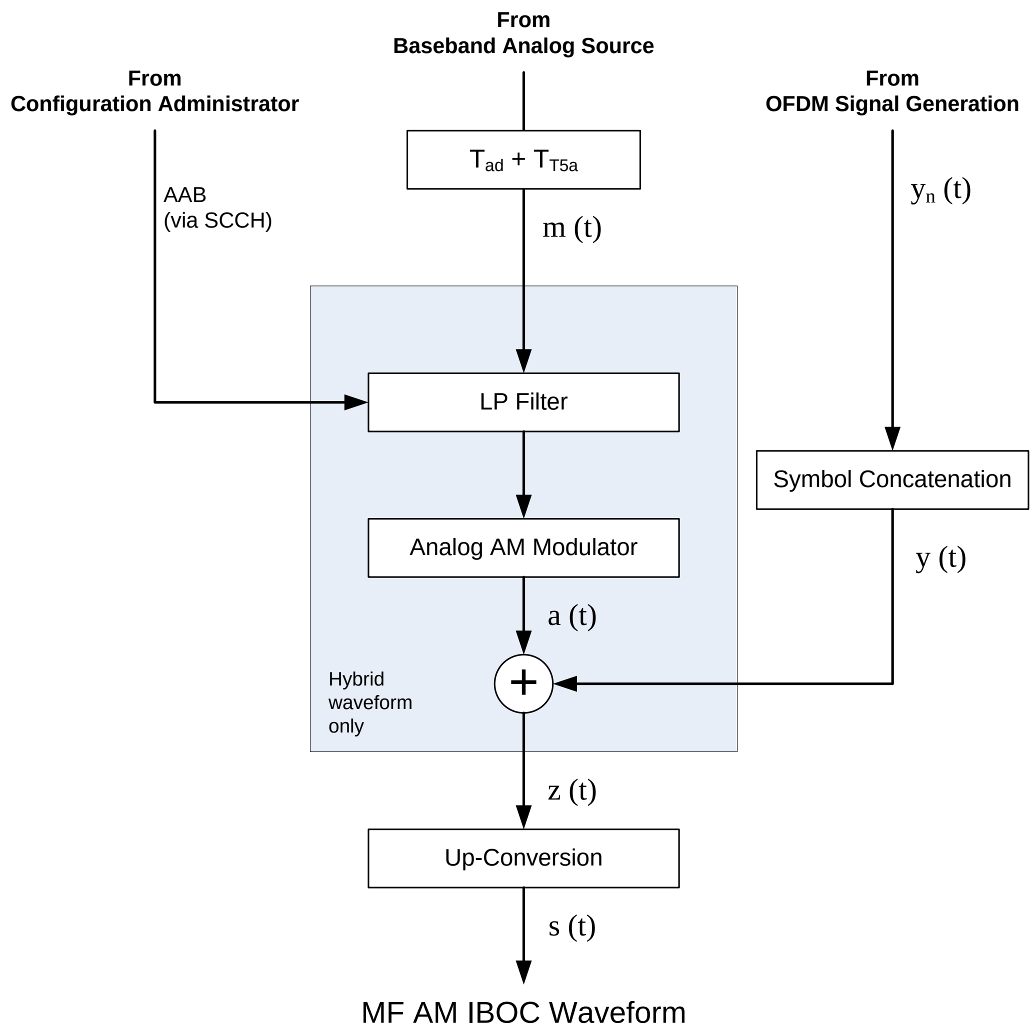

Figure 14-1: Hybrid Transmission Subsystem Functional Block Diagram

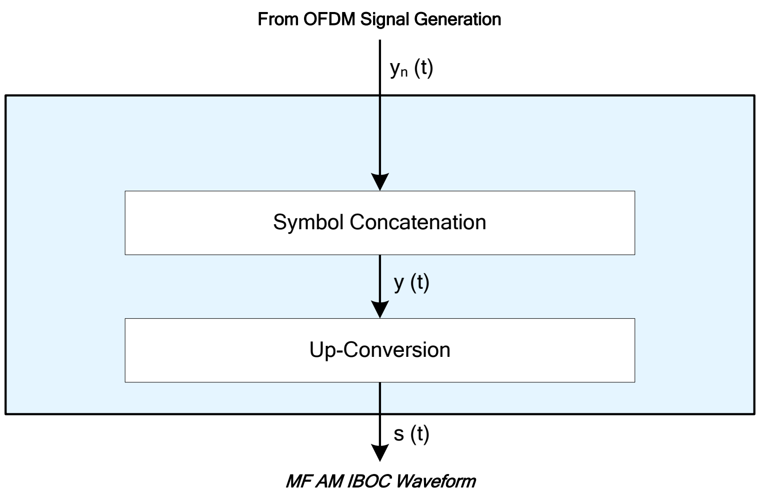

Figure 14-2: All Digital Transmission Subsystem Functional Block Diagram

14.2 Functional Components

The functional components of the transmission subsystem are specified in Subsections 14.2.1 through 14.2.6.

14.2.1 Symbol Concatenation

The individual time-domain OFDM symbols are summed to produce a continuum of pulses over $0 \le t \le \infty$ as follows:

$$y(t) = \sum_{n=0}^{\infty} y'_n(t)$$

14.2.2 Low-Pass Filtering

Low-pass filtering of the analog audio source, m(t), is necessary when broadcasting the Hybrid waveform in order to limit interference to the digital subcarriers from the analog host. For the standard digital bandwidth configuration (RDB = 0), the bandwidth of this filter depends on the state of the AAB bit received from the Configuration Administrator. If the AAB bit is zero, the analog audio shall be filtered to a 5 kHz bandwidth according to the 5 kHz specifications in [13]. If the AAB bit is one, the analog audio shall be filtered to an 8 kHz bandwidth according to the 8 kHz specifications in [13].

For the reduced digital bandwidth configuration (RDB = 1), the analog audio shall be filtered to a 9.4 kHz bandwidth according to the 9.4 kHz specifications in [13].

This low-pass filtering can be done in external audio processors also.

14.2.3 Analog Diversity Delay

When broadcasting the Hybrid waveform, the digital signal is combined with the analog AM signal, as shown in Figure 14-1. However, analog diversity delay is first applied to the baseband analog AM signal.

In the HD Radio system, the analog and digital signals carry the same audio program with the analog audio delayed from the corresponding digital audio at the output of the analog/digital combiner. This delay consists of a fixed portion T_{ad}, as defined in Subsection 3.5 (AM System Parameters), plus an adjustable portion T_{T5a}. The delay is adjusted so that the audio content in the analog and digital paths has a time diversity of precisely T_{ad} at the TX antenna. This delay accounts for processing delay differences in the two signal paths.

The absolute accuracy of the analog diversity delay, when enabled, is defined in [9].

Ball-game mode: A radio station can disable the analog diversity delay for specialized broadcasts. The state of the analog diversity delay is indicated by the Blend Control bits in the Audio Transport layer (See Reference [4]). However, changing the state of the analog diversity delay may result in a discontinuity during reception as the receiver blends from analog to digital. Some receivers may disable digital reception entirely when analog diversity delay is disabled.

14.2.4 Analog AM Modulator

When broadcasting the Hybrid waveform, this process computes the envelope of the analog AM signal by applying a modulation index and adding a DC offset as follows:

$$a(t) = [1 + g \cdot m(t)]$$

where a(t) is the envelope, m(t) is the analog source, and g is the modulation gain. Typically, g = 1.25, representing a +125% modulation level. The input analog audio source, m(t), must be preprocessed external to the AM HD Radio Exciter so that a(t) does not assume negative values. See Reference [9] for a complete description of the requirements on the input analog audio source.

In addition, the analog signal will be compatible with the Emergency Alert System (EAS) as specified in Title 47 CFR Part 11 (See Reference [11]). The analog spectral emissions mask, per Title 47 CFR §73.317 [12], is contained in Reference [9].

14.2.5 Analog/Digital Combiner

When broadcasting the Hybrid waveform, the real analog AM baseband waveform, a(t), is coherently combined with the digital baseband waveform, y(t), to produce the complex baseband AM HD Radio Hybrid waveform z(t), as follows:

$$\mathrm{Re}\,[z(t)] = \mathrm{Re}\,[y(t)] + a(t)$$

$$\mathrm{Im}\,[z(t)] = \mathrm{Im}\,[y(t)]$$

The levels of the digital sidebands in the output spectrum are appropriately scaled by OFDM subcarrier mapping (see Section 12). These scale factors are provided in [9].

Changing service modes from any Hybrid service mode to any other Hybrid service mode (including future backward compatible Hybrid service modes) shall not cause any interruptions or discontinuities in the analog signal. Refer to [9] for further details.

14.2.6 Up-Conversion

The concatenated digital signal z(t) is translated from baseband to the RF carrier frequency as follows:

$$s(t) = \mathrm{Re}\left(e^{j2\pi f_c t} \cdot z(t)\right)$$

where f_c is the RF channel frequency and Re( ) denotes the real component of the complex quantity. For the All Digital waveform, z(t) is replaced with y(t).

The AM HD Radio waveform is broadcast in the current AM radio band and its power levels and spectral content [9] are limited to be within the spectral mask as defined in 47 C.F.R §73.44.

The carrier frequency spacing and channel numbering schemes are compatible with 47 CFR §73.14. Channels are centered at 10 kHz intervals ranging from 540 to 1700 kHz. Both the analog and digital portions of the Hybrid waveform are centered on the same carrier frequency. The absolute accuracy of the carrier frequency is defined in [9].

15 GLOSSARY

For the purpose of better understanding this document, the following definitions apply:

Absolute L1 Frame Number (ALFN) - A number assigned to each transmitted L1 frame that provides a reference to absolute time. The start of ALFN 0 occurred at 00:00:00 Coordinated Universal Time (UTC) on January 6, 1980. The start of every subsequent L1 frame occurs at an exact integer multiple of T_f after that instant in time.

All Digital waveform - The transmitted waveform composed of digitally modulated primary, secondary, and tertiary OFDM subcarriers. Use of this waveform may be preceded by an interim phase using the Hybrid waveform. The All Digital waveform is the more robust transmission medium. (See Hybrid waveform.)

allocated channel - One of the 117 possible frequency assignments in the AM band, as defined in Reference [12].

amplitude modulation (AM) - Modulation in which the amplitude of a carrier wave is varied in accordance with the level of the modulating signal.

amplitude scale factor - A factor which multiplies the baseband components of a particular OFDM subcarrier of the transmitted spectrum to constrain the radiated power to a prescribed level.

analog audio bandwidth control (AAB) - A parameter that indicates which of two audio bandwidths are to be transmitted in the Hybrid mode. AAB is not applicable in the reduced digital bandwidth configuration (RDB = 1).

analog signal - Refers to signals that are modulated on the main carrier by conventional continuously varying amplitude modulation. (See digital signal.)

BC range - The range of L1 Blocks, m1:m2, spanned by a transfer frame, indicating its position within an L1 frame.

bit mapping - The last step in the interleaving process. Assigns each subframe bit to a row and column in an interleaver matrix as well as to a unique bit position in the digital word within the interleaver matrix element. This reordering of bits before transmission mitigates the impact of burst errors caused by signal fades and interference.

BPSK (Binary Phase Shift Keying) - A form of digital phase modulation that assigns one of two discrete phases, differing by 180 degrees, to the carrier. Each BPSK symbol conveys one bit of information.

channel encoding - The process used to add redundancy to each of the logical channels to improve the reliability of the transmitted information.

characterization parameters - The unique set of defining parameters (transfer, latency, and robustness) for each logical channel for a given service mode. The channel encoding, interleaving, spectral mapping, and diversity delay of the logical channel determine its characterization parameters.

code rate - Defines the increase in overhead on a coded channel resulting from channel encoding. It is the ratio of information bits to the total number of bits after coding.

convolutional encoding - A form of forward error correction channel encoding that inserts coding bits into a continuous stream of information bits to form a predictable structure. Unlike a block encoder, a convolutional encoder has memory; its output is a function of current and previous inputs.

configuration administrator - The configuration administrator is a system function that configures each of the layers using SCCH information or parameters which do not change often.

digital signal - Refers to signals that are digitally modulated on subcarriers by OFDM. (See analog signal.)

diversity delay - Imposition of a relative time delay between two channels carrying the same information to defeat non-stationary channel impairments such as fading and impulsive noise.

fading - The variation (with time) of the amplitude or relative phase (or both) of one or more frequency components of a received signal.

frequency modulation (FM) - Modulation in which the instantaneous frequency of a sine wave carrier is caused to depart from the channel center frequency by an amount proportional to the instantaneous amplitude of the modulating signal.

grounded conductive structures - Metal structures connected to earth ground such as towers and bridges that can attenuate and/or re-radiate the MF radio signal.

Hybrid waveform - The transmitted waveform composed of the analog AM signal, plus digitally modulated primary and optionally secondary and tertiary OFDM subcarriers. This waveform supports operation of both analog and digital receivers and may be used in an interim phase preceding conversion to the All Digital waveform. (See All Digital waveform.)

interleaver block - A logical subdivision of an interleaver partition. Each interleaver block contains 32 rows and C columns (where C = 25 or C = 2).

interleaver depth - The number of rows in an interleaver matrix. The system employs two interleaver depths: L1 block (32 rows) and L1 frame (256 rows).

interleaver matrix - A two-dimensional array used to reorder subframe bits. The AM system uses five different interleaver matrices.

interleaving - A reordering of the message bits to distribute them in time (over different OFDM symbols) and frequency (over different OFDM subcarriers) to mitigate the effects of signal fading and interference.

L1 block - A unit of time of duration T_b. Each L1 frame is comprised of 8 L1 blocks.

L1 block count - An index that indicates one of 8 equal subdivisions of an L1 frame.

L1 block rate - The rate, equal to the reciprocal of the L1 block duration, $\left(\frac{1}{T_b}\right)$, at which selected transfer frames are conducted through Layer 1.

L1 frame - A specific time slot of duration T_f identified by an ALFN. The transmitted signal may be considered to consist of a series of L1 frames.

L1 frame rate - The rate, equal to the reciprocal of the L1 frame duration, $\left(\frac{1}{T_f}\right)$, at which selected transfer frames are conducted through Layer 1.

latency - The time delay that a logical channel imposes on a transfer frame as it traverses Layer 1. One of the three characterization parameters. (See robustness and transfer.)

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 several higher layer services into logical channels (partitioned into L1 frames and L1 blocks) for processing in Layer 1.

L2 protocol data units (PDU) - Units of user content and upper layer protocol control information transferred from Layer 2 to Layer 1. (See PDU.)

logical channel - A signal path that conducts transfer frames from Layer 2 through Layer 1 with a specified grade of service.

lower sideband - The group of OFDM subcarriers (subcarriers number -1 through -81) below the carrier frequency.

mother code - The complete code sequence generated by a convolutional encoder. (See puncturing.)

OFDM signal generation - The function that generates the modulated baseband signal in the time domain.

OFDM subcarrier - A discrete frequency-domain signal within the allocated channel that encodes digital data through its amplitude and/or phase. The total set of subcarriers, taken in aggregate for a period of T_s, provides the digital data for that time interval. (See OFDM symbol.)

OFDM subcarrier mapping - The function that assigns the interleaved logical channels to the OFDM subcarriers.

OFDM symbol - Time-domain pulse of duration T_s, representing all the active subcarriers and containing all the data in one row of all the interleaver matrices and a bit from the system control data sequence vector. The transmitted waveform is the concatenation of successive OFDM symbols.

Orthogonal Frequency Division Multiplexing (OFDM) - A parallel multiplexing scheme that modulates a data stream onto a large number of orthogonal subcarriers that are transmitted simultaneously. (See OFDM symbol.)

parity - In binary-coded data, a condition maintained so that in any permissible coded expression, the total number of “1”s or “0”s is always odd, or always even.

power level control (PL) - In the Hybrid waveform, the nominal level of the secondary, PIDS, and tertiary sidebands (relative to the analog carrier) is one of two selectable values: low or high. Power level control (PL) specifies which level is to be employed, where PL=0 selects the low level and PL=1 selects the high level.

primary sidebands - The OFDM sidebands consisting of subcarriers 57 through 81 and -57 through -81 with the Hybrid waveform and subcarriers 2 through 26 and -2 through -26 with the All Digital waveform.

pulse-shaping function - A time-domain pulse superimposed on the OFDM symbol to improve its spectral characteristics.

puncturing - The process of removing selected bits from the mother codeword to increase FEC code rate.

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.

QPSK (Quadrature Phase Shift Keying) - A form of digital phase modulation that assigns one of four discrete phases, differing by 90 degrees, to the carrier. Each QPSK symbol conveys two bits of information.

reduced digital bandwidth control (RDB) - A parameter that is used to limit the spectrum of the MA1 and MA3 waveforms. In service mode MA1, RDB may be used to reduce interference to the analog host and accommodate an analog bandwidth of up to 9.4 kHz. In service mode MA3, RDB may be used reduce interference to adjacent channels

robustness - The ability of a logical channel to withstand channel impairments such as noise, interference, and fading. There are eight distinct levels of robustness designed into Layer 1 of the AM air interface. One of the three characterization parameters. (See latency and transfer.)

scrambling - The process of modulo-2 summing the input data bits with a pseudo-random bit stream to randomize the time-domain bit stream.

secondary sidebands - The OFDM sidebands consisting of subcarriers 27 through 53 and -27 through -53 in the Hybrid mode and subcarriers 27 through 52 in the all digital mode.

service mode - A specific configuration of operating parameters specifying throughput, performance level, and selected logical channels.

service mode control - Control information passed over the SCCH from the Configuration Administrator to Layer 1 which determines the service mode for Layer 1.

signal constellation mapper - The process in OFDM subcarrier mapping that associates sets of bits with specific 64-QAM, 16-QAM, QPSK, or BPSK states.

spectral emissions mask - A specification setting the maximum level of out-of-band components of the transmitted signal.

spectral mapping - The association of specific logical channels with specific subcarriers or groups of subcarriers.

subframe generation - The first step in the interleaving process. Splits the bits of each active input transfer frame into multiple subframes and, at the same time, reorders the bits. The number of subframes for an active logical channel, the division and order of transfer frame bits, and the number of transfer frames needed to fill the subframes, are all dependent upon service mode.

system control - Data from the Configuration Administrator conveying control such as service mode, power level, analog audio bandwidth, and analog diversity delay.

system control channel (SCCH) - A channel which transports control information from the Configuration Administrator to Layer 1 and also conveys status information from Layer 1 to Layer 2, through the system control processing.

system control data sequence - A sequence of bits destined for each reference subcarrier representing the various system control components relayed between Layer 1 and Layer 2.

system control processing - The function that generates the system control data sequence.

symbol concatenation - The process of concatenating individual OFDM symbol pulses in time to produce a continuous time-domain signal.

system protocol stack - The protocols associated with operation of the various functional layers.

system time alignment, T_{st} - Internal time delay to absorb variations in internal processing time to maintain message alignment with L1 frames and blocks.

tertiary sidebands - The OFDM sidebands consisting of subcarriers 2 through 26 and -2 through -26 with the Hybrid Waveform and subcarriers -27 through -52 with the All Digital waveform.

training - A known pattern or sequence of bits, the training pattern or training sequence, is intermingled in the transmitted data sequence to allow the receiver to detect and correct for the effects of non-uniform channel effects over the transmission path and receiver front end.

transfer - A measure of the data throughput through a logical channel. One of the three characterization parameters. (See latency and robustness.)

transfer frame - An ordered, one-dimensional collection of data bits of specified length grouped for processing through a logical channel for exchange with the physical layer.

transfer frame number - A number, $F^{n}_{m1:m2}$, that specifies the ALFN, n, and BC range, m1:m2, associated with a particular transfer frame, in order to relate the transfer frame to absolute time.

transfer frame rate - The number of transfer frames per second

transfer frame size - The number of bits in a transfer frame.

transmission subsystem - The functional component used to format and up-convert the baseband HD Radio waveform for transmission through the medium frequency (MF) channel.

transmit time alignment, T_{T1a} - Adjusted so that the digital time diversity between main and backup is precisely T_{dd} at the transmit antenna.

transmit audio alignment, T_{T5a} - Adjusted so that the audio content in the analog and digital paths has a time diversity of precisely T_{ad} at the TX antenna (applies to hybrid service modes only)

upper sideband - The group of OFDM subcarriers (subcarriers number +1 through +81) above the carrier frequency.

vector - A one-dimensional array.

HD Radio™ Air Interface Design Description Layer 2 Channel Multiplex

Version K

November 2022

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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 defines Layer 2: the channel multiplexer. Specific hardware and software implementations are not described. See References [1], [2], [7], [9], [4], [5], and [6] for more details.

2 Reference Documents

STATEMENT
Each referenced document that is mentioned in this document shall be listed in the following iBiquity document:

- Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard Document Number: SY_REF_2690s

3 Abbreviations and Conventions

3.1 Introduction

Section 3 provides the following:

- Abbreviations and Acronyms
- Presentation Conventions
- Mathematical Symbols

3.2 Abbreviations and Acronyms

AAS	Advanced Application Services
AAT	AAS Data Transport
ALFN	Absolute Layer 1 Frame Number
AM	Amplitude Modulation
BC	L1 Block Count
CA	Conditional Access
CW	Control Word
DDL	Data Delimiter
FM	Frequency Modulation
GF	Galois Field
HD RLS	HD Radio Link Subsystem
IBOC	In-Band On-Channel
ISO	International Organization for Standardization
L1	Layer 1
L2	Layer 2
LLDS	Low-Latency Data Service
LSB	Least Significant Bit
MF	Medium Frequency
MPS	Main Program Service
MPSA	Main Program Service Audio
MPSD	Main Program Service Data
MSB	Most Significant Bit
PCI	Protocol Control Information
PDU	Protocol Data Unit
PIDS	Primary IBOC Data Service Logical Channel
PIDSOV	Primary IBOC Data Service Overlay Logical Channel
POV	Primary Overlay Logical Channel
PSD	Program Service Data
RDB	Reduced Digital Bandwidth
RS	Reed Solomon
SIDS	Secondary IBOC Data Service Logical Channel
SIS	Station Information Service
SPS	Supplemental Program Service
SPSA	Supplemental Program Service Audio
SPSD	Supplemental Program Service Data
VHF	Very High Frequency

3.3 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.

3.4 Mathematical Symbols

3.4.1 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 component 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	$\underline{\underline{u}}[i][j]$, $\underline{\underline{V}}[i][j]$
$\begin{bmatrix} & & \\ & & \end{bmatrix}$	Indicates the components 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]

3.4.2 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
CEIL( )	Indicates the least integer greater than or equal to the real number	CEIL(6.432) = 7
$\oplus$	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	$\log_{10}(100) = 2$
*	Indicates complex conjugate	If x = (3 + j4), x* = (3 - j4)
0x	Indicates a hexadecimal value	0x10 = 16

4 Layer 2 Transport – Description

The primary function of Layer 2 is to receive audio and data from various higher layers within the HD Radio system, multiplex this information into Layer 2 Protocol Data Units (PDUs) and route these PDUs to the appropriate Layer 1 logical channel. The data received from the higher layers is also in the form of PDUs but from the individual transport layers providing the service. Layer 2 enables the HD Radio system to support five transport services as described below and shown in Figure 4-1:

1. Main Program Service (MPS) which includes Main Program Service Audio (MPSA) and may also include Main Program Service Data (MPSD). MPS PDUs are generated by the Audio Transport and encapsulate both MPSA and MPSD information.
2. Supplemental Program Service (SPS) provides the broadcaster the option of multiplexing additional programs with the MPS. The SPS includes Supplemental Program Service Audio (SPSA) and may also include Supplemental Program Service Data (SPSD). SPS PDUs are generated by the same Audio Transport as the MPS PDUs.
3. Advanced Application Services (AAS) allow the broadcaster to multiplex additional types of content other than MPS and SPS. The AAS transport provides the packet transport mechanism for these data services. It performs the framing and the encapsulation of the data packets. There are two types of methods for multiplexing AAS data into a Layer 2 PDU: fixed and opportunistic. Fixed data is granted a fixed bandwidth allocation by purposely scaling back the bandwidth allocation of the MPS and/or SPS, whereas opportunistic makes use of any unused bandwidth due to variability of both the MPS and SPS Audio. Many service modes do not allow blending between analog audio and MPS digital audio; in some of these modes, it is possible that neither MPS audio nor SPS audio, but only AAS data, is present in the Layer 2 PDUs.
4. The Station Information Service (SIS, for AM and FM) is a specialized transport/data link for transmitting SIS data on the Primary IBOC Data Service (PIDS, PIDSOV) and the Secondary IBOC Data Service (SIDS) Layer 1 logical channels.
5. The Low-Latency Data Service (LLDS, for FM only) is a specialized transport/data link for transmitting LLDS data on the PIDS, PIDSOV, and SIDS Layer 1 logical channels.

For SIS and LLDS, Layer 2 does not perform a multiplexing function, but simply passes the PDUs directly into the Layer 1 PIDS, PIDSOV, and/or SIDS logical channels without additional overhead in the form of headers. SIS and LLDS are the only services carried by the PIDS, PIDSOV, and SIDS logical channels. SIS and LLDS are defined in References [6] and [37], respectively.

The HD Radio system supports various configurations with respect to Layer 1. Based on the Layer 1 service mode, the system provides multiple Layer 1 logical channels. The number of active Layer 1 logical channels and the characteristics defining them vary for each service mode. The defining characteristics of each Layer 1 logical channel are:

- Transfer Frame Size
- Transfer Frame Rate
- Robustness
- Latency

Details of the logical channels used for each L1 service mode are described in References [1] and [2].

With respect to the exchange between Layer 2 and Layer 1, Layer 2 is subjected to the Layer 1 configuration and timing. The configuration is governed by the control information received from the Configuration Administrator. The total Layer 1 frame size consists of audio/data content from upper protocol layers, Reed Solomon (RS) parity (if present), and the L2 Protocol Control Information (PCI) overhead. Layer 2 allows the MPS/SPS and AAS Transports to be active within any active Layer 1 logical channel, with the exception of PIDS, PIDSOV, and SIDS.

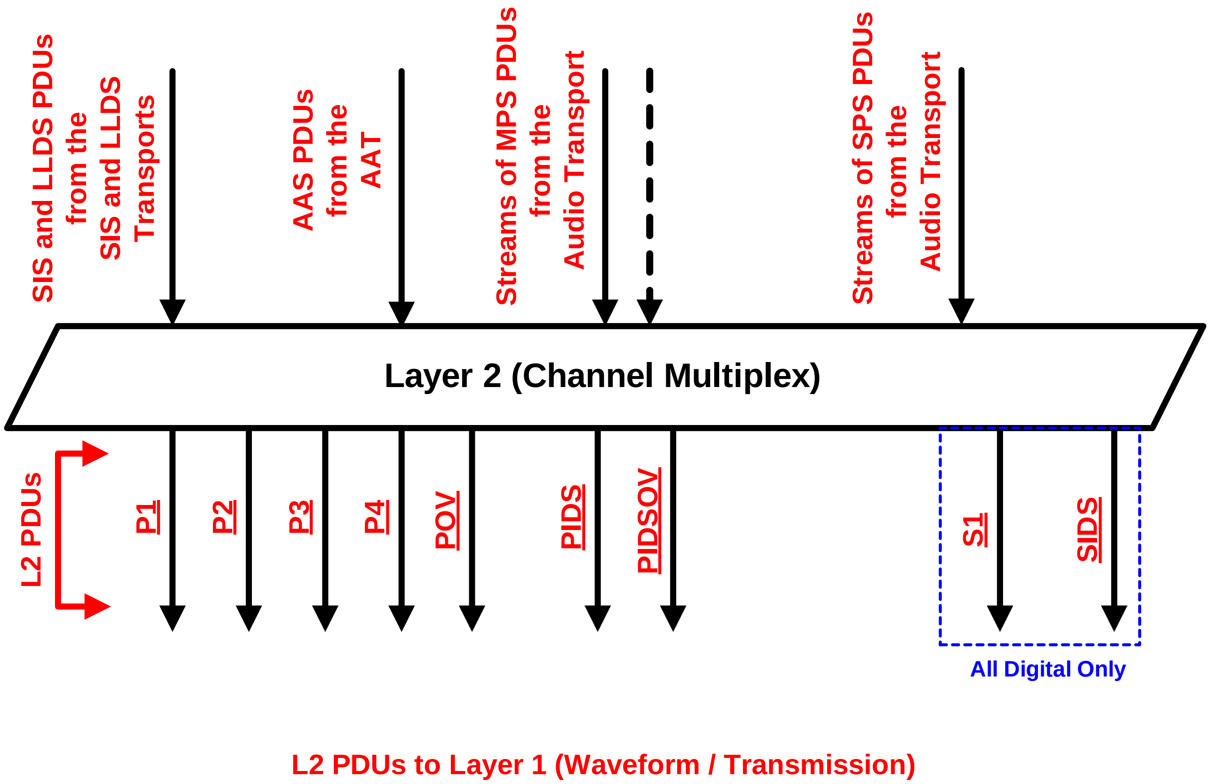

Figure 4-1: Layer 2 – Interface Diagram

In addition to the Layer 2 PDUs, status information is also passed between Layer 1 and Layer 2. The status information passed from Layer 1 to Layer 2 consists of *Absolute L1 Frame Number* (ALFN) and *L1 Block Count* (BC).

5 Layer 2 PDU Generation

This section describes the structure, content, and generation of Layer 2 PDUs. It defines the Layer 2 Protocol Control Information (PCI) included as part of the Layer 2 PDU for most Layer 1 logical channels in the HD Radio system. It also describes the details of how the various service PDUs are multiplexed into a Layer 2 PDU, and the Reed Solomon (RS) coding applied to some L2 PDUs.

5.1 Layer 2 PDU Structure and Content

There are two types of L2 PDUs: standard and advanced. AM service modes MA1 and MA3, and standard FM service modes MP1, MP2, MP3, MP5, MP6, and MP11, convey only standard L2 PDUs. Advanced FM service modes DSB1, DSB1OV, and MS5 convey only advanced L2 PDUs. Advanced FM service modes MP1X, MP1XOV, and MP6OV convey both standard and advanced L2 PDUs. Logical channels in advanced L2 PDUs are characterized by special modulation, coding, and diversity techniques that improve their robustness and/or capacity.

Table 5-1 indicates whether the logical channels in each service mode are carried by standard or advanced L2 PDUs. Since Layer 2 simply passes SIS and LLDS PDUs directly into Layer 1 PIDS, PIDSOV, and/or SIDS logical channels without adding any overhead, those logical channels are not included in Table 5-1.

Table 5-1: L2 PDU Types

Service Mode	Logical Channel	L2 PDU Type
MA1	P1	Standard
	P3	Standard
MA3	P1	Standard
	P3	Standard
MP1	P1	Standard
MP2	P1	Standard
	P3	Standard
MP3	P1	Standard
	P3	Standard
MP5	P1	Standard
	P2	Standard
	P3	Standard
MP6	P1	Standard
	P2	Standard
MP11	P1	Standard
	P3	Standard
	P4	Standard
MP1X	P1	Standard
	P4	Advanced
DSB1	P1	Advanced
MP1XOV	P1	Standard
	P4	Advanced
	POV	Advanced
MP6OV	P1	Standard
	P2	Standard
	POV	Advanced

Service Mode	Logical Channel	L2 PDU Type
DSB1OV	P1	Advanced
	POV	Advanced
MS5	S1	Advanced

The structure of standard L2 PDUs is shown in Figure 5-1, and the structure of advanced L2 PDUs is shown in Figure 5-2. L2 PDUs may contain various combinations of MPSA, MPSD, SPSA, SPSD, opportunistic data, and fixed data. They may also contain encoded PCI bits to identify this content. In standard L2 PDUs, the encoded PCI bits are distributed evenly throughout the PDU. In advanced L2 PDUs, the PCI is more robustly encoded and positioned for simplicity in a single field at the end of the PDU. Advanced L2 PDUs also carry RS parity bytes to improve the robustness of the audio and data content. Finally, a data delimiter (DDL) marks the opportunistic data boundary within both standard and advanced PDUs.

SPS, AAS, and (in some cases) MPS are optional. However, when available, the structure of a Layer 2 PDU can contain five different possible combinations of audio and data:

a. The payload is audio oriented (MPS/SPS) only.
b. A mixed content payload, containing MPS/SPS, opportunistic data.
c. A mixed content payload, containing MPS/SPS, fixed data.
d. A mixed content payload, containing MPS/SPS, opportunistic data and fixed data.
e. The payload contains fixed data only.

Figure 5-3 shows various ways in which audio and data content can be arranged within an L2 PDU. This does not apply to the PIDS, PIDSOV, or SIDS logical channels, which carry SIS and possibly LLDS PDUs.

Opportunistic data is made available only when the audio (MPS/SPS) does not use its allocated bandwidth. The MPS/SPS PDU lengths are based on the maximum bit rate for a particular audio codec mode. The unused portions of the bandwidth are then aggregated and used to include opportunistic data. It can originate in both the MPS and SPS; however, it is combined in the AAS Data Transport using the HD RLS before sending it to Layer 2 as part of the AAS PDUs.

Thus, opportunistic data is PDU-specific and cannot be guaranteed at any particular rate or instance in time, making it a service of lesser quality. Also, fixed data and opportunistic data can occur independently across logical channels.

L2 PDUs containing fixed AAS data include an additional extended header within the HD RLS. The format and structure of both fixed and opportunistic data processed by the HD RLS is described in [5]. The mixed content fixed/opportunistic PDU requires additional indications. To allow opportunistic data to be identified in the Layer 2 PDU, the 5-byte DDL field is used to identify the start of the opportunistic data in the PDU. Refer to [5] for details.

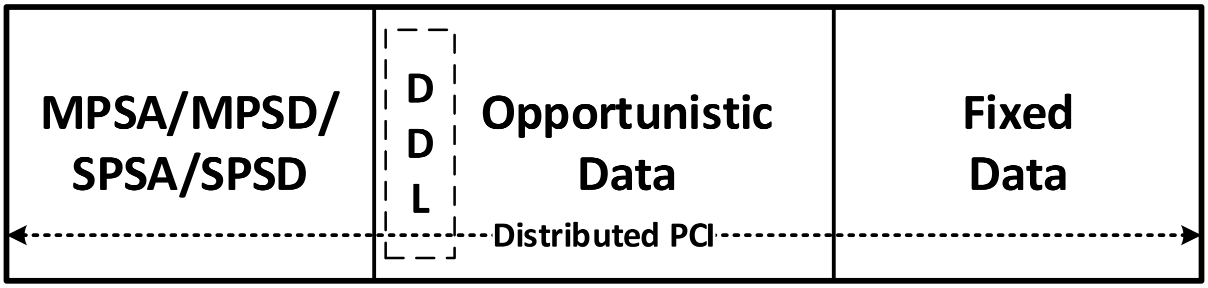

Figure 5-1: Standard L2 PDU Structure

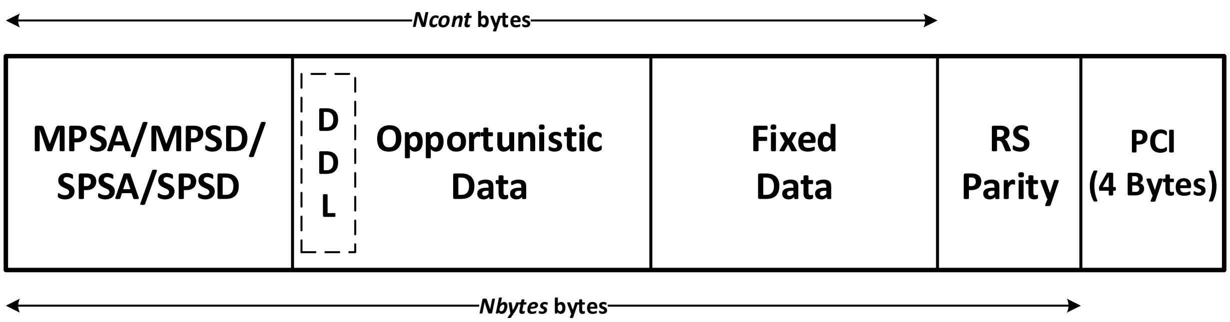

Figure 5-2: Advanced L2 PDU Structure

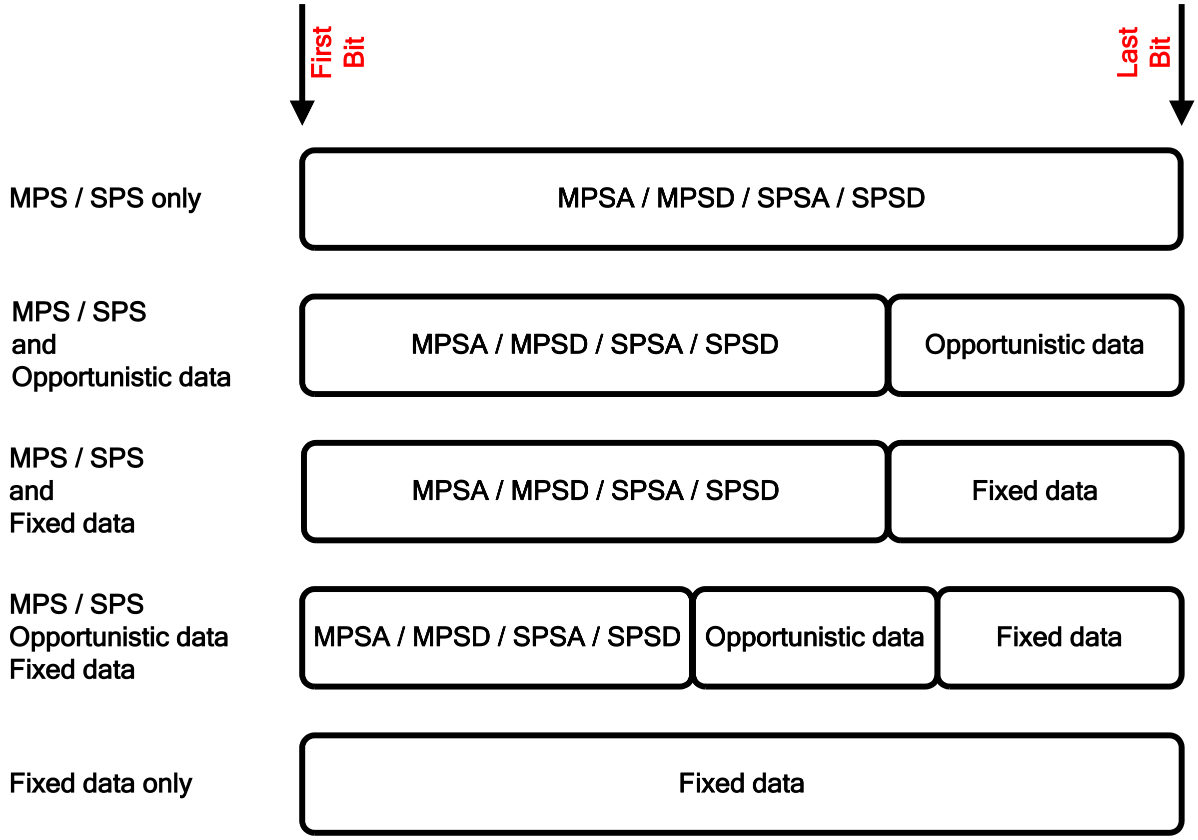

Figure 5-3: L2 PDU Content Arrangements

5.1.1 MPS, SPS, and AAS Multiplexing

MPS and SPS employ an identical transport mechanism; thus, care must be taken when multiplexing these PDUs into a Layer 2 PDU so that the receiver can correctly process the PDUs as well as provide the listener with an accurate description of the available programs. This requires further attention because MPSA can contain core and enhanced streams that are transported on different Layer 1 logical channels. This subsection describes the restrictions and configurations when multiplexing and processing MPS and SPS programs, as well as AAS data.

Table 5-2 shows the allowable mapping of MPS, SPS, and AAS services to Layer 1 logical channels for all standard and advanced AM and FM service modes. Logical channels that are bolded show required mappings of service to logical channel, and those that are not bolded show optional mappings. To maintain compatibility with the existing base of fielded receivers, the P1 logical channel must always carry core MPS audio in service modes MA1 and MA3. Likewise, the P1 logical channel must always carry single-stream MPS audio in service modes MP1, MP2, MP3, MP11, MP1X, and MP1XOV.

Non-blend hybrid service modes MP5, MP6, DSB1, MP6OV, and DSB1OV may carry digital content that is unrelated to the analog audio. This digital content may be audio-only, data-only, or a mixture of both audio and data. If MPS digital audio is broadcast using these service modes, diversity delay will not be applied to the analog audio, even if the content is identical. MPS digital audio in service modes MP5, MP6, and MP6OV shall consist of core audio carried on the P1 logical channel, with optional enhanced audio on the P2 logical channel.

Table 5-2: Allowable Mapping of MPS, SPS, and AAS Services to Logical Channels

Service Mode(s)	MPS	SPS	AAS
MP1	**P1**	P1	P1
MP2	**P1**	P1, P3	P1, P3
MP3	**P1**	P1, P3	P1, P3
MP11	**P1**	P1, P3, P4	P1, P3, P4
MP5	P1 (core) P2 (enhanced)	P2, P3	P1, P2, P3
MP5 + MS5	P1 (core) P2 (enhanced)	P2, P3, S1	P1, P2, P3, S1
MP6	P1 (core) P2 (enhanced)	P1, P2	P1, P2
MP6 + MS5	P1 (core) P2 (enhanced)	P1, P2, S1	P1, P2, S1
MP1X	**P1**	P1, P4	P1, P4
DSB1	P1	P1	P1
DSB1 + MS5	P1, S1	P1, S1	P1, S1
MP1XOV	**P1**	P1, P4, POV	P1, P4, POV
MP6OV	P1 (core) P2 (enhanced)	P1, P2, POV	P1, P2, POV
MP6OV + MS5	P1 (core) P2 (enhanced)	P1, P2, POV, S1	P1, P2, POV, S1
DSB1OV	P1, POV	P1, POV	P1, POV
DSB1OV + MS5	P1, POV, S1	P1, POV, S1	P1, POV, S1
MA1	**P1 (core)** P3 (enhanced)	P3	P1, P3

Service Mode(s)	MPS	SPS	AAS
MA1 (Reduced Digital Bandwidth)	**P1 (core)**	None	P1
MA3	**P1 (core)** P3 (enhanced)	P3	P1, P3
MA3 (Reduced Digital Bandwidth)	**P1 (core)**	None	P1

5.1.2 FM Example Configurations

Table 5-3 shows various example configurations for mapping services/programs to logical channels for each standard FM service mode. The desired broadcast configuration is chosen based on the quality of service for the particular application, in combination with allocation requirements as described in this document. Table 5-3 is a set of sample configurations based on a maximum of three SPSs in addition to the MPS; however, additional configurations could be added to include more SPSs. Note that in service modes MP5 and MP6, MPS and SPS audio programs may not be present.

Table 5-4 shows various example configurations for mapping services/programs to logical channels for each advanced FM service mode. Due to the increased capacity of advanced FM service modes, sample configurations for up to eight audio programs are shown. Except for MP6OV, advanced service modes carry only single-stream audio programs (i.e., they do not support core and enhanced audio streams). Furthermore, in advanced FM service modes DSB1, DSB1OV, and MP6OV it is possible that MPS and SPS audio may not be present at all. The PIDS, PIDSOV, and SIDS logical channels may carry SIS and/or LLDS data.

Layer 2 PDU construction adheres to the following guidelines:

- The Main Program Service is always designated as Program Number "0".
- SPS1 through SPS7 can reference Program Numbers 1 through 7, respectively. Refer to [4] for a detailed description on Program Number and program indications.
- AAS Data consists of fixed data and opportunistic data, if available.
- Programs 1 through 7 can be added or removed at any time. Program 0 is constantly present in standard service modes MP1, MP2, MP3, and MP11, and backward-compatible advanced service modes MP1X and MP1XOV.
- The bit rate of MPS enhanced audio streams may be reduced to 0 kbps.
- SPS programs must use single-stream audio.
- Hybrid service modes MP5, MP6, DSB1, MP6OV, and DSB1OV may carry only data services.
- The core audio stream shall always be added or removed first in a multi-stream program.
- The Main Program Service shall always be first in the order of the physical L2 PDU.
- Free-access Supplemental Programs can be placed in any order in the PDU.
- Conditionally-accessed Supplemental Programs shall be placed last in the PDU. Refer to Subsection 5.3 for details.

Table 5-3: Mapping of Services/Programs to Logical Channels for Standard FM Service Modes – Example Configurations

Service Mode	MPS		SPS-1	SPS-2	SPS-3	SIS Data	AAS Data (Fixed and Opportunistic)
	Core Channel	Enhanced Channel					
MP1	**P1**	—	—	—	—	PIDS	—
	P1	—	P1	—	—	PIDS	P1
	P1	—	P1	P1	—	PIDS	—
MP2	**P1**	—	—	—	—	PIDS	P3
	P1	—	P3	—	—	PIDS	P1
	P1	—	P1	P3	—	PIDS	—
MP3	**P1**	—	—	—	—	PIDS	P3
	P1	—	P3	—	—	PIDS	P1
	P1	—	P1	P3	—	PIDS	P1
	P1	—	P1	—	—	PIDS	P3
	P1	—	P1	P1	P3	PIDS	—
MP11	**P1**	—	—	—	—	PIDS	P3, P4
	P1	—	P1	P3	—	PIDS	P4
	P1	—	P1	P1	P4	PIDS	P3
	P1	—	P1	P3	P4	PIDS	P1
MP5	—	—	—	—	—	PIDS	P1, P2, P3
	P1	P2	P3	—	—	PIDS	—
	P1	P2	P2	P3	—	PIDS	—
	P1	P2	P2	—	—	PIDS	P3
	P1	P2	P2	P3	—	PIDS	P2
	P1	P2	P3	—	—	PIDS	P2
	P1	—	P2	P2	P3	PIDS	P2
MP6	—	—	—	—	—	PIDS	P1, P2
	P1	P2	—	—	—	PIDS	—
	P1	P2	P2	—	—	PIDS	P1
	P1	P2	P1	—	—	PIDS	P2
	P1	—	P2	P2	—	PIDS	P1
	P1	P2	—	—	—	PIDS	P2
	P1	P2	P1	P2	—	PIDS	—

NOTE: Logical channels that are **bolded** show required mappings because MPS audio must be supported in these service modes.

Table 5-4: Mapping of Services/Programs to Logical Channels for Advanced FM Service Modes – Example Configurations

Service Mode	MPS	SPS-1	SPS-2	SPS-3	SPS-4	SPS-5	SPS-6	SPS-7	SIS / LLDS Data	AAS Data (Fixed and Opportunistic)
MP1X	**P1**	P4	—	—	—	—	—	—	PIDS	P4
	P1	P1	P4	P4	—	—	—	—	PIDS	P1
	P1	P1	P1	P4	P4	—	—	—	PIDS	—
DSB1	—	—	—	—	—	—	—	—	PIDS	P1
	P1	P1	P1	P1	—	—	—	—	PIDS	P1
	P1	P1	P1	P1	P1	P1	—	—	PIDS	—
	P1	P1	P1	P1	P1	P1	P1	P1	PIDS	P1
MP1XOV	**P1**	P4	POV	—	—	—	—	—	PIDS/ PIDSOV	POV
	P1	P4	POV	POV	—	—	—	—	PIDS/ PIDSOV	P1
	P1	P1	POV	POV	POV	—	—	—	PIDS/ PIDSOV	P4
	P1	P1	P1	P4	POV	POV	POV	—	PIDS/ PIDSOV	—
MP6OV	—	—	—	—	—	—	—	—	PIDS/ PIDSOV	P1, P2, POV
	P1(c)/ P2(e)	POV	—	—	—	—	—	—	PIDS/ PIDSOV	POV
	P1(c)/ P2(e)	P2	POV	POV	—	—	—	—	PIDS/ PIDSOV	P1
	P1(c)/ P2(e)	P1	POV	POV	POV	—	—	—	PIDS/ PIDSOV	P2
	P1(c)	P2	P2	POV	POV	POV	—	—	PIDS/ PIDSOV	—
DSB1OV	—	—	—	—	—	—	—	—	PIDS/ PIDSOV	P1, POV
	P1	P1	POV	POV	—	—	—	—	PIDS/ PIDSOV	P1
	P1	P1	P1	POV	POV	—	—	—	PIDS/ PIDSOV	P1
	P1	P1	P1	P1	POV	POV	—	—	PIDS/ PIDSOV	POV
	POV	P1	P1	P1	P1	POV	POV	—	PIDS/ PIDSOV	—
MP5 + MS5	—	—	—	—	—	—	—	—	PIDS/ SIDS	P1, P2, P3, S1
	P1(c)/ P2(e)	P3	S1	S1	—	—	—	—	PIDS/ SIDS	—
	P1(c)/ P2(e)	P2	P3	—	—	—	—	—	PIDS/ SIDS	S1
	P1(c)/ P2(e)	P2	S1	S1	—	—	—	—	PIDS/ SIDS	P3
	P1(c)/	P2	P3	S1	S1	S1	—	—	PIDS/	P2

Service Mode	**MPS**	**SPS-1**	**SPS-2**	**SPS-3**	**SPS-4**	**SPS-5**	**SPS-6**	**SPS-7**	**SIS / LLDS Data**	**AAS Data (Fixed and Opportunistic)**
	P2(e)								SIDS	
	P1(c)/ P2(e)	P3	—	—	—	—	—	—	PIDS/ SIDS	S1
	P1(c)	P2	P2	P3	—	—	—	—	PIDS/ SIDS	S1
MP6 + MS5	—	—	—	—	—	—	—	—	PIDS/ SIDS	P1, P2, S1
	P1(c)/ P2(e)	S1	S1	—	—	—	—	—	PIDS/ SIDS	—
	P1(c)/ P2(e)	P2	S1	S1	—	—	—	—	PIDS/ SIDS	P1
	P1(c)/ P2(e)	P1	S1	S1	S1	—	—	—	PIDS/ SIDS	P2
	P1(c)	P2	P2	S1	—	—	—	—	PIDS/ SIDS	S1
	P1(c)/ P2(e)	—	—	—	—	—	—	—	PIDS/ SIDS	S1
	P1(c)/ P2(e)	P1	P2	S1	—	—	—	—	PIDS/ SIDS	S1
DSB1 + MS5	—	—	—	—	—	—	—	—	PIDS/ SIDS	P1, S1
	P1	P1	P1	P1	—	—	—	—	PIDS/ SIDS	S1
	P1	P1	P1	P1	P1	P1	S1	S1	PIDS/ SIDS	—
	S1	P1	P1	P1	P1	S1	—	—	PIDS/ SIDS	P1
MP6OV + MS5	—	—	—	—	—	—	—	—	PIDS/ SIDS/ PIDSOV	P1, P2, POV, S1
	P1(c)/ P2(e)	POV	S1	S1	—	—	—	—	PIDS/ SIDS/ PIDSOV	POV
	P1(c)/ P2(e)	P2	POV	POV	S1	S1	—	—	PIDS/ SIDS/ PIDSOV	P1
	P1(c)/ P2(e)	P1	POV	POV	POV	S1	S1	S1	PIDS/ SIDS/ PIDSOV	P2
	P1(c)	P2	P2	POV	POV	POV	—	—	PIDS/ SIDS/ PIDSOV	S1
	P1(c)/ P2(e)	POV	POV	POV	S1	S1	S1	—	PIDS/ SIDS/ PIDSOV	—
DSB1OV +	—	—	—	—	—	—	—	—	PIDS/ SIDS/ PIDSOV	P1, POV, S1

Service Mode	MPS	SPS-1	SPS-2	SPS-3	SPS-4	SPS-5	SPS-6	SPS-7	SIS / LLDS Data	AAS Data (Fixed and Opportunistic)
MS5	P1	P1	POV	POV	S1	S1	—	—	PIDS/ SIDS/ PIDSOV	P1
	P1	P1	P1	POV	POV	—	—	—	PIDS/ SIDS/ PIDSOV	S1
	S1	P1	P1	P1	POV	POV	S1	—	PIDS/ SIDS/ PIDSOV	POV
	POV	P1	P1	P1	P1	POV	S1	S1	PIDS/ SIDS/ PIDSOV	—

NOTE: P1(c) refers to P1 Core; P2(e) refers to P2 Enhanced. Logical channels that are **bolded** show required mappings because MPS audio must be supported in these service modes.

Logical channels may transport encoded MPS audio on the core stream or on both the core stream and the enhanced stream. Since the core stream must be present before the enhanced stream can be added, removing the core stream effectively removes the specific program. See Reference [4] for more information on the core and enhanced bit streams and the nominal bit rates.

The core audio stream is sent over the more robust logical channel. Refer to [1] for a detailed description of service modes, logical channels, and backward compatibility.

For the Main Program Service, the core stream is always provided over logical channel P1. Providing the core stream over logical channel P1 is necessary to ensure backward compatibility of service modes.

For MPS Audio programs, core and enhanced streams shall not both be on the same logical channel. Similarly, a particular stream shall not be split across logical channels. SPS programs are always single-stream.

5.1.3 AM Example Configurations

Table 5-5 shows various example configurations for mapping services/programs to logical channels for each AM service mode. The desired broadcast configuration is chosen based on the quality of service for the particular application, in combination with allocation requirements as described in this document.

Table 5-5 shows the standard MA1 configuration of core and enhanced MPS audio. The core and/or enhanced audio bit rates can be scaled back to provide AAS data on the P1 and/or P3 logical channels. MA1 MPS enhanced audio on the P3 logical channel can also be disabled and replaced with an SPS audio program. In this case, the SPS audio on the P3 channel is single-stream. There is insufficient capacity in service mode MA1 to simultaneously support an SPS with acceptable audio quality and AAS data.

Similar configurations are shown for MA3. Sufficient capacity exists in this service mode to allow SPS audio on the P3 logical channel and AAS data on both P1 and P3 logical channels.

In the MA1 and MA3 reduced digital bandwidth configurations (RDB = 1), the secondary and tertiary subcarriers are not transmitted, and all capacity is carried on the P1 logical channel. In this case, MPS audio is a single core stream which can optionally be scaled back in bit rate to provide AAS data.

Table 5-5: Mapping of Services/Programs to Logical Channels for AM Service Modes – Example Configurations

Service Mode	Reduced Digital Bandwidth (RDB)	MPS		SPS-1	SIS Data	AAS Data (Fixed and Opportunistic)
		Core Channel	Enhanced Channel			
MA1	0	**P1**	P3	—	PIDS	P1
	0	**P1**	P3	—	PIDS	P3
	0	**P1**	P3	—	PIDS	P1, P3
	0	**P1**	—	P3	PIDS	—
	1	**P1**	—	—	PIDS	P1
MA3	0	**P1**	P3	—	PIDS	P1
	0	**P1**	P3	—	PIDS	P3
	0	**P1**	P3	—	PIDS	P1, P3
	0	**P1**	—	P3	PIDS	P1, P3
	1	**P1**	—	—	PIDS	P1

Note: RDB = 1 indicates that information is carried by only the P1 logical channel. Logical channels that are **bolded** show required mappings because MPS audio must be supported in these service modes.

Layer 2 PDU construction adheres to the following guidelines:

- The Main Program Service is always designated as Program Number “0”.
- SPS1 is always designated as Program Number “1”.
- AAS Data consists of fixed data and opportunistic data, if available.
- SPS1 can be added or removed at any time. MPS shall always be present.
- The bit rate of MPS enhanced audio streams may be reduced to 0 kbps.
- SPS1 programs must use single-stream audio.
- The core audio stream shall always be added or removed first in a multi-stream program.
- The Main Program Service shall always be first in the order of the physical L2 PDU.

5.2 Layer 2 PCI

The PCI identifies the contents of an L2 PDU using information obtained from the Configuration Administrator. This information consists of:

- A flag that indicates whether an L2 PDU contains an MPS PDU
- The maximum size allocated for the MPS PDU
- A flag that indicates the presence of one or more SPS PDUs
- A flag that indicates whether an L2 PDU contains AAS Data PDU(s)
- The maximum size allocated for AAS Data PDU(s)

The encoding and location of the PCI depends on the L2 PDU type. In standard L2 PDUs, the encoded PCI bits are distributed evenly throughout the PDU. In advanced L2 PDUs, the PCI is encoded differently and positioned in a single field at the end of the PDU.

5.2.1 PCI for Standard L2 PDUs

The encoded PCI, or header bits, for standard L2 PDUs consists of one of eight cyclic permutations, CW_0 through CW_7, of a 24-bit sequence. The encoded PCI sequences and the corresponding indication types are described in Table 5-6.

The contents of a selected codeword are designated as $[h_0, h_1, h_2, \ldots, h_{21}, h_{22}, h_{23}]$.

Table 5-6: Generic Header Sequence Indications

Sequence	Binary Header Sequence $[h_0, h_1, h_2, \ldots, h_{21}, h_{22}, h_{23}]$	Hexadecimal Equivalent	MPSA/MPSD/ SPSA/SPSD	Fixed Data	Opportunistic Data
CW_0	[001110001101100011010011]	0x38D8D3	Yes	No	No
CW_1	[110011100011011000110100]	0xCE3634	Yes	No	Yes
CW_2	[111000110110001101001100]	0xE3634C	Yes	Yes	No
CW_3	[100011011000110100110011]	0x8D8D33	Yes	Yes	Yes
CW_4	[001101100011010011001110]	0x3634CE	No	Yes	No
CW_5	[100011010011001110001101]	0x8D338D	Reserved	Reserved	Reserved
CW_6	[110110001101001100111000]	0xD8D338	Reserved	Reserved	Reserved
CW_7	[011000110100110011100011]	0x634CE3	Reserved	Reserved	Reserved

To improve robustness, the header bits are evenly spread over most of the Layer 2 PDU, as shown in Figure 5-4. The payload is quantified in units of bytes. Any excess payload that does not constitute a byte is located at the end of the payload. The h_0 header bit is offset from the beginning of the transfer frame by N_{start} bits. Each remaining header bit is separated from the previous header bit by N_{offset} bits. N_{offset} refers to the number of bits between each pair of header bits, exclusive of the header bits themselves.

These numbers depend on the L2 PDU length (in bits), L, as shown in Table 5-7. If the L2 PDU length is an integral number of bytes, the header length is 24 bits. If the L2 PDU length is not an integral number of bytes, the header is shortened to either 23 or 22 bits as shown. If the header length is 23 bits, then h_{23} is not used. If the header length is 22 bits, then h_{22} and h_{23} are not used.

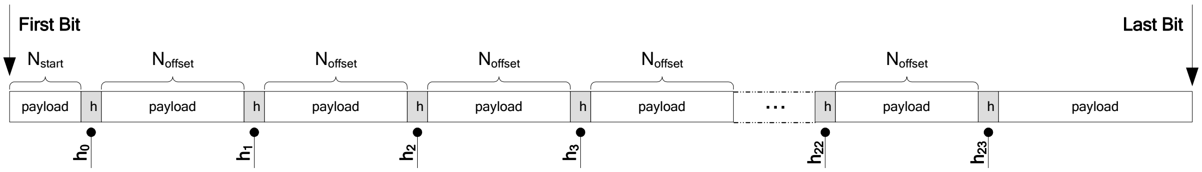

Figure 5-4: Standard L2 Transfer Frame

Table 5-7: Header Spread Parameters

L2 PDU Length, L (Bits)	(L MOD 8) =	N_{start} (Bits)	N_{offset} (Bits)	Header Length (Bits)
< 72000	0	120	$8 \cdot INT\left\{\frac{CEIL\left[\frac{(L-120)}{8}\right]}{24}\right\} - 1$	24
	7	120	$8 \cdot INT\left\{\frac{CEIL\left[\frac{(L-120)}{8}\right]}{23}\right\} - 1$	23
	1-6	120	$8 \cdot INT\left\{\frac{CEIL\left[\frac{(L-120)}{8}\right]}{22}\right\} - 1$	22
≥ 72000	0	$8 \cdot CEIL\left[\frac{(L-30{,}000)}{8}\right]$	1247	24
	7	$8 \cdot CEIL\left[\frac{(L-30{,}000)}{8}\right]$	1303	23
	1 - 6	$8 \cdot CEIL\left[\frac{(L-30{,}000)}{8}\right]$	1359	22

5.2.2 PCI for Advanced L2 PDUs

The changes to the PCI in the advanced L2 PDU include the use of a more robust PCI encoding technique and replacing the distributed PCI bits with a 4-byte PCI field located at the end of the advanced L2 PDU.

The content information from the Configuration Administrator can be configured into a row vector of PCI identification bits:

$$PCI\_ID = \begin{pmatrix} b_5 & b_4 & b_3 & b_2 & b_1 & b_0 \end{pmatrix}$$

This vector can be more compactly represented using only the three LSBs (b2, b1, b0), with three spare identification bits (b5, b4, b3) allowing for future expansion. The three LSBs indicate the presence of the three possible forms of content in an L2 PDU: MPSA/MPSD/SPSA/SPSD, Opportunistic Data, and Fixed Data, with various subsets permissible, as shown in Table 5-8.

Table 5-8: PCI Content Identification for Advanced L2 PDUs

PCI Code-word	**PCI Identification Bits b_2, b_1, b_0**	**PCI Field [$f_{31}, f_{30}, f_{29}, \ldots, f_2, f_1, f_0$]**	**MPSA/MPSD/ SPSA/SPSD**	**Fixed Data**	**Opportunistic Data**
CW_0	0 0 0	[00000000000000000000000000000000] = 0x00000000	Yes	No	No
CW_1	0 0 1	[01010101010101010101010101010101] = 0x55555555	Yes	No	Yes
CW_2	0 1 0	[00110011001100110011001100110011] = 0x33333333	Yes	Yes	No
CW_3	0 1 1	[01100110011001100110011001100110] = 0x66666666	Yes	Yes	Yes
CW_4	1 0 0	[00001111000011110000111100001111] = 0x0F0F0F0F	No	Yes	No
CW_5	1 0 1	[01011010010110100101101001011010] = 0x5A5A5A5A	Reserved	Reserved	Reserved
CW_6	1 1 0	[00111100001111000011110000111100] = 0x 3C3C3C3C	Reserved	Reserved	Reserved
CW_7	1 1 1	[01101001011010010110100101101001] = 0x 69696969	Reserved	Reserved	Reserved

5.3 Order of Content

The mix of content from different sources, each having a separate transport mechanism, requires mapping that can be tracked by the receivers under various channel conditions or under various operating scenarios. In addition, established rules for mapping content allow for future introduction of services while maintaining backward compatibility.

The structure of a Layer 2 PDU, as shown in Figure 5-3, indicates the relative placement of audio and data content. For a Layer 2 PDU, the left-most bit is referred to as bit zero (bit 0) and the right-most bit is referred to as MSB. When audio content, opportunistic data, and fixed data are present, the audio content is placed from bit 0, followed by opportunistic data, and then followed by fixed data.

A more inclusive description, as shown in Figure 5-5, emphasizes the distribution of the protocol control information (PCI) across the complete standard L2 PDU bit aggregate. The PCI bits are independent of the payload content and are (logically) placed first, before the rest of the PDU is constructed.

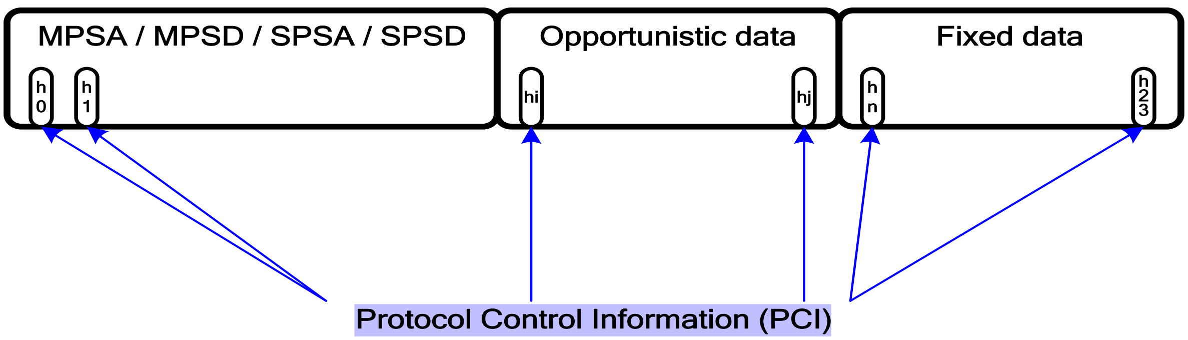

Figure 5-5: Order of Content for Standard L2 PDU

Audio-oriented content may include MPS transport and multiple instances of SPS transport. Figure 5-6 provides a detailed description of certain order requirements within the audio content in Layer 2.

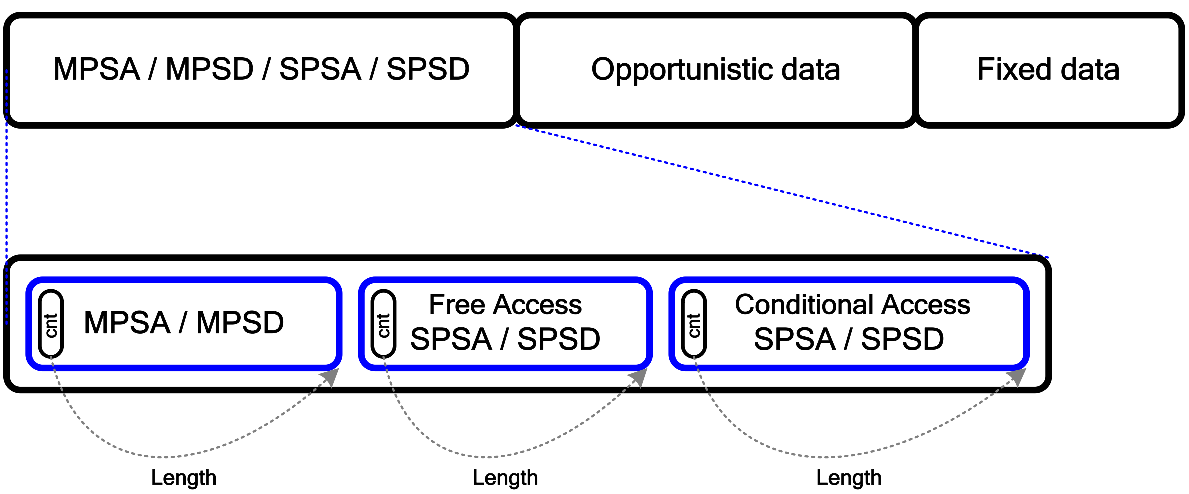

Figure 5-6: Layer 2 Order of Multiple Instances of Audio-Oriented Content

MPS transport is always placed first, starting with Layer 2 bit 0. MPS transport is then followed by instances of SPS transport that contain audio that is free-access. Instances of SPS transport that contain audio that is conditionally-accessed are placed last, toward the end of the audio-oriented content. The audio-oriented transport instances are independent. Each instance contains its control information that points to the end of the PDU. Therefore, the control information of the last audio instance points to the end of the entire audio-oriented content.

AAS data-oriented content (for services) may include opportunistic data services transport and fixed data services transport. Figure 5-7 provides a detailed description of certain order requirements within the AAS data services content in Layer 2. Note that if there is no opportunistic data being sent, the DDL field shall not be sent either.

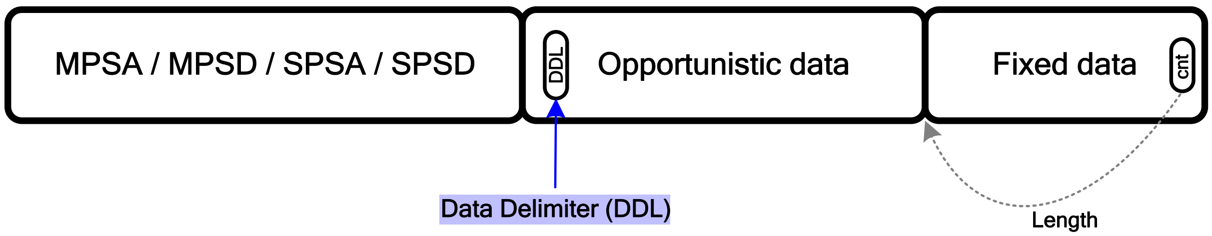

Figure 5-7: Layer 2 Order of Data Content with Control Information

The fixed data services transport PDU is always placed towards the last bit of the standard L2 PDU, or just prior to the RS parity field in the advanced L2 PDU. The fixed data transport control information, which is included within the fixed data PDU, points to the end of that PDU.

The opportunistic data PDU immediately follows (counting down bitwise) the fixed data PDU. It may span any instantaneous length and does not necessarily fill the entire gap (i.e., it does not necessarily utilize the instantaneously available bandwidth) towards the audio content. The end of the opportunistic data PDU (lower end bitwise count) is indicated by a Data Delimiter (DDL).

5.4 Concatenated RS Encoding for Advanced L2 PDUs

Advanced L2 PDUs are comprised of multiple systematic RS codewords to afford additional error detection and correction capability. The RS parity bytes, located in the RS Parity field of the advanced L2 PDU as shown in Figure 5-2, are generated from the audio/data content of the PDU. The computed RS parity bytes for all RS codewords are grouped together into a single parity field and appended to the existing L2 PDU content, as shown in Figure 5-8.

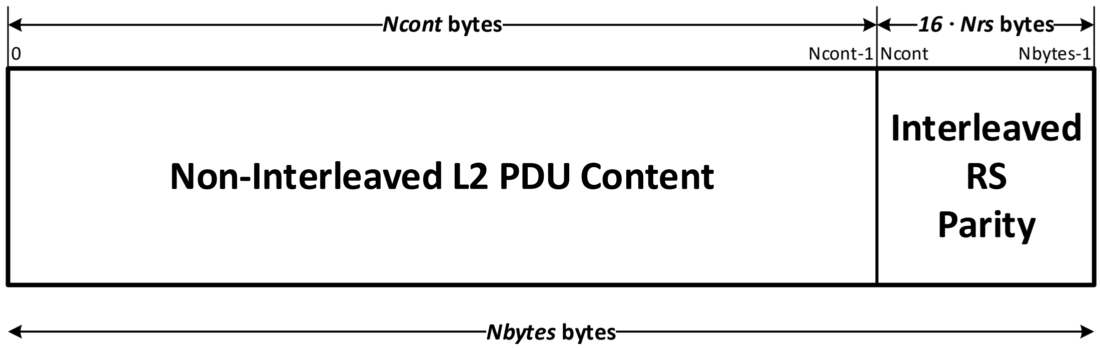

Figure 5-8: Advanced L2 PDU Content and Parity

The number of bytes available for L2 PDU content (MPSA/MPSD/SPSA/SPSD, Opportunistic Data, and Fixed Data) and RS parity is denoted as Nbytes, as shown in Figure 5-8. The RS Parity field, consisting of 16 parity bytes per RS codeword, is generated after first defining the advanced service mode (e.g., DSB1) to determine the value of Nbytes for each advanced L2 PDU in that service mode. The following expressions are used to define the RS encoding process:

- Number of bytes per advanced L2 PDU, excluding PCI: $Nbytes$
- Number of RS codewords per advanced L2 PDU: $Nrs = ceil\left(\frac{Nbytes}{255}\right)$
- Number of content bytes per advanced L2 PDU: $Ncont = Nbytes - 16 \cdot Nrs$
- Number of bytes per RS codeword: $RSbytes = ceil\left(\frac{Nbytes}{Nrs}\right)$

The concatenated encoding process begins with prepending the content field with zero-padding, as shown in Figure 5-9, to prepare it for interleaving prior to RS encoding.

- Number of prepended zero bytes: $NPzero = Nrs \cdot RSbytes - Nbytes$

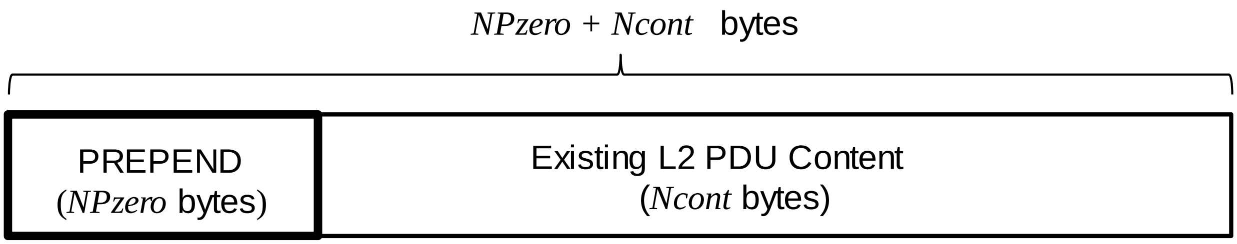

Figure 5-9: Prepended Content Field (Pcontent) Used Only for RS Interleaving and Encoding

Prepending and subsequent interleaving of the L2 PDU content creates an integer multiple of equal-length RS codewords. These codewords are then input to an RS encoder to generate 16 bytes of parity for each codeword. The parity bytes are then interleaved and appended, in a single parity field, to the end of the original (neither prepended nor interleaved) L2 PDU content. Thus, prepending and interleaving of the content are required only for parity computation; they do not affect the transmission of the actual content. This process is described in detail below.

The number of bytes in the prepended content field Pcontent is an integer multiple of the number of RS codewords Nrs. Interleaving then segments the prepended content into Nrs systematic components of equal size (RSbytes-16). The systematic portion of the RS codewords (Pcontent) is formatted using the following interleaver expression:

$$RSsys_{n,k} = Pcontent_{k+n \cdot Nrs} \quad ; \quad n = 0 \ldots RSbytes - 17, \; k = 0 \ldots Nrs - 1$$

where RSsys is a matrix of the Pcontent (systematic) bytes of the codewords. There are Nrs column vectors, one for each RS codeword. The interleaving enabled by RSsys is used only to generate the parity fields, as the existing advanced L2 PDU content is transmitted without interleaving.

Each column vector of RSsys is then input to a Reed Solomon encoder RS(RSbytes,RSbytes-16,GF(28)) to generate a 16-byte parity result RSP for each of the Nrs codewords. These parity results are themselves interleaved to form the RS Parity field RSpar, assembled using the expression:

$$RSpar_{k+n \cdot Nrs} = RSP_{n,k} \quad ; \quad n = 0 \ldots 15, \; k = 0 \ldots Nrs - 1$$

where $RSP_{n,k}$ is the nth byte (n=0…15) of the kth (k=0…Nrs-1) parity result RSP. The RS Parity field RSpar is then appended to the end of the original L2 PDU content, and followed by the PCI field, to form the advanced L2 PDU shown in Figure 5-2.

The 16 parity bytes for each RS codeword are generated by dividing the RS systematic component by the RS generator polynomial over $GF(2^8)$, and taking the remainder of the polynomial division. The RS codewords are considered “shortened” since the number of bytes RSbytes in each codeword is generally less than 255 for $GF(2^8)$. Note that any prepended zero bytes at the start of the RSsys codewords do not change the polynomial division parity results (RSP) for the shortened codewords. The purpose of prepending is to facilitate convenient interleaving of equal-size RS codewords.

The RS code is a standard code defined over $GF(2^8)$. This code is defined by its primitive polynomial and generator polynomial.

The primitive polynomial is:

$$p(x) = x^8 + x^4 + x^3 + x^2 + 1$$

or 100011101 in binary notation, where the LSB is on the right. The generator polynomial is:

$$g_{16}(x) = a^{136} + a^{240}x + a^{208}x^2 + a^{195}x^3 + a^{181}x^4 + a^{158}x^5 + a^{201}x^6 + a^{100}x^7 + a^{11}x^8 + a^{83}x^9 + a^{167}x^{10} + a^{107}x^{11} + a^{113}x^{12} + a^{110}x^{13} + a^{106}x^{14} + a^{121}x^{15} + x^{16}$$

where “a” is a root of the primitive polynomial.

The logical channels in advanced FM service modes that are carried by advanced L2 PDUs are listed in Table 5-1. The RS encoding parameters for these logical channels are defined in Table 5-9. Note that PIDS, SIDS, and PIDSOV logical channels are not included in Table 5-9 because they are passed without modification directly into Layer 1 from SIS and LLDS PDUs.

Table 5-9: Summary of RS Encoding Parameters for Advanced L2 PDUs

Advanced Service Mode	**Logical Channel**	***Nbytes***	***Nrs***	***Ncont***	***NPzero***	***RSbytes***
MP1X	P4	1724	7	1612	5	247
DSB1	P1	5676	23	5308	5	247
MP1XOV	P4	860	4	796	0	215
	POV	2836	12	2644	8	237
MP6OV	POV	2836	12	2644	8	237
DSB1OV	P1	5676	23	5308	5	247
	POV	2836	12	2644	8	237
MS5	S1	2836	12	2644	8	237

6 Layer 2 Processing

6.1 Transmit Processing Description

For each active Layer 1 logical channel, Layer 1 indicates to Layer 2 that it requires an L2 PDU. Based on the parameters defined in the previous section, L2 indicates to the Audio Transport and the AAS Data Transport (AAT) to provide their respective PDUs (MPS, SPS PDU, and AAS PDU) that are to be multiplexed within the L2 PDU to that specific Layer 1 logical channel.

Once Layer 2 has received the input PDUs, it creates the L2 PDU to be sent to the appropriate Layer 1 logical channel by:

- Creating Layer 2 PCI based on content and encoding
- Spreading the PCI across a standard L2 PDU or appending it to the end of an advanced L2 PDU, as defined in Table 5-1
- Generating RS parity bytes and placing them prior to the PCI bytes in advanced L2 PDUs
- Inserting MPS/SPS and AAS PDUs at the start of the L2 PDU

The upper layers inform L2 what information is available to it.

For a PIDS, SIDS, or PIDSOV Layer 1 logical channel, Layer 1 indicates to Layer 2 that it requires an L2 PDU. Layer 2 requests the SIS or LLDS Transport to provide the respective PDU. Layer 2 forwards the SIS or LLDS PDU directly to Layer 1 without any modification.

The HD Radio system provides SIS to all HD Radio receivers. The PIDS, SIDS, and PIDSOV logical channels transport SIS that may be acquired quickly for initial screening of provided services and other station-related information.

HD Radio™ Air Interface Design Description Audio Transport

Version I

November 2022

SY_IDD_1017s

TRADEMARKS

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1 Scope

1.1 System Overview

iBiquity Digital Corporation's 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 not described.

2 Reference Documents

STATEMENT
Each referenced document that is mentioned in this document shall be listed in the following iBiquity document:

- Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard Document Number: SY_REF_2690s

3 Abbreviations and Conventions

3.1 Introduction

Section 3 provides the following:

- Abbreviations and Acronyms
- Presentation Conventions
- Mathematical Symbols

3.2 Abbreviations and Acronyms

AAS	Advanced Application Services
AF	Audio Frame
AM	Amplitude Modulation
C	Center
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
MPS	Main Program Service
MPSA	Main Program Service Audio
MPSD	Main Program Service Data
N/A	Not Applicable
NOP	Number of Packets
PCI	Protocol Control Information
PCM	Pulse Code Modulation
PDU	Protocol Data Unit
PSD	Program Service Data
R	Right
RBDS	Radio Broadcast Data System
RS	Reed-Solomon
RX	Receiver
SPS	Supplemental Program Service
SPSA	Supplemental Program Service Audio
SPSD	Supplemental Program Service Data
VHF	Very High Frequency

3.3 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.

3.4 Mathematical Symbols

3.4.1 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	$\underline{\underline{u}}[i][j]$, $\underline{\underline{V}}[i][j]$
$\begin{bmatrix} & \\ & \end{bmatrix}$	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

3.4.2 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

3.4.3 Data Constant Formats

The data constant formats defined below are used throughout this document.

Category	Definition	Examples
0b	Indicates a binary number	0b1111 = 15 (decimal)
0x	Indicates a hexadecimal number	0xFF = 255 (decimal)

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. Each audio program has its own individual PDU.

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 encoded audio packets, generates PDUs in the Audio Transport, and conveys the 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 [10] and multiplexes this data with encoded audio. Thus, the output streams contain both encoded audio and PSD. Main Program Service Data (MPSD) provides additional information about the Main Program Service Audio (MPSA) and Supplemental Program Service Data (SPSD) provides additional information about 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 [5], thus allowing for the inclusion of opportunistic data. The definitions of the units used for data transfer – audio frames, encoded audio packets, PDUs – within the Audio Transport layer are explained in the Glossary.

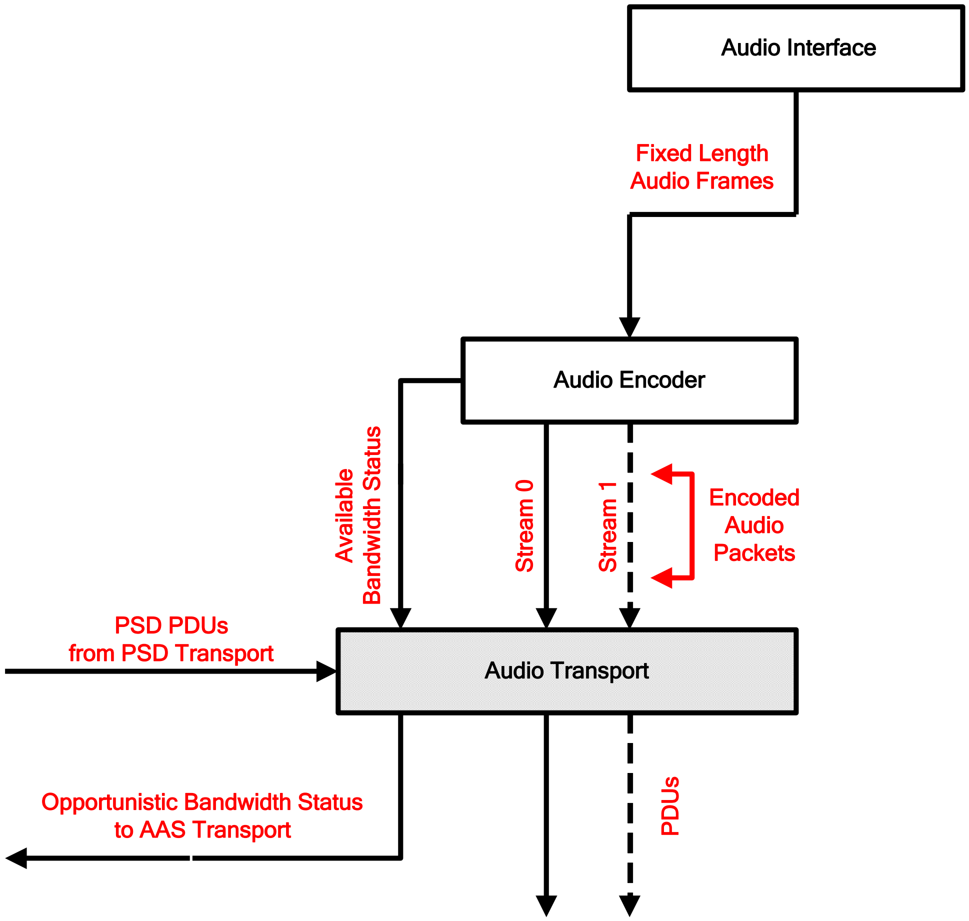


Figure 4-1: Audio Transport Interface Diagram

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.

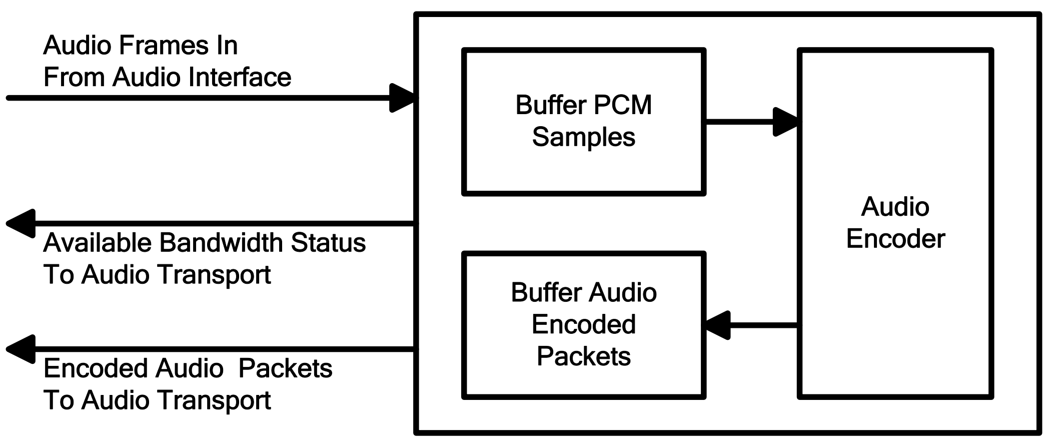


Figure 4-2: Transmit Audio Encoder Interface

The Audio Encoder is flexible in format and bit rate scaling. It provides high quality, efficient audio compression over a variety of formats from 8.1-kbit/s for a monophonic channel to 96-kbit/s for a stereo audio format. The audio transport also provides a fixed rate Program Service Data byte-stream channel and supports variable rate opportunistic data channel.

4.3 Audio Transport

The Audio Transport accepts the variable length encoded 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 packets per PDU will be within 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 (frames), with an average of 32 packets (frames) per PDU. 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 encoded audio packets is variable to accommodate extra throughput per packet. The parameter D is sent over-the-air as part of the 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.

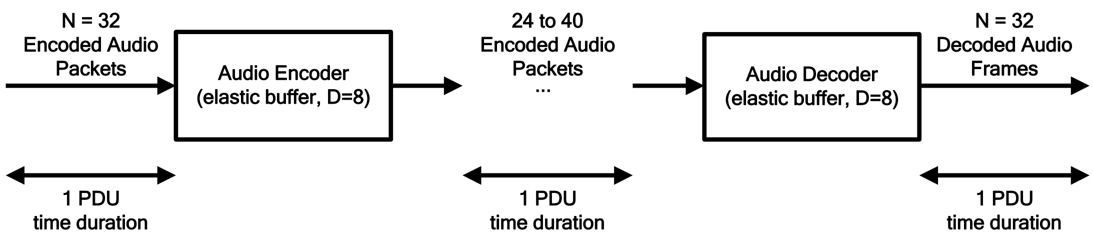


Figure 4-3: Elastic Buffering Example

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 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 enhanced stream. In this example, a valid enhanced PDU will be generated only once for every eight valid core PDUs.

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

At the receiver, the encoded audio 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 audio packets. This assumes that the rate control mechanism for the codec variability is implemented within the Decoder. In addition, appropriate buffering of encoded audio packets is implemented by the Audio Transport such that a constant delay is maintained through the encode/decode process.

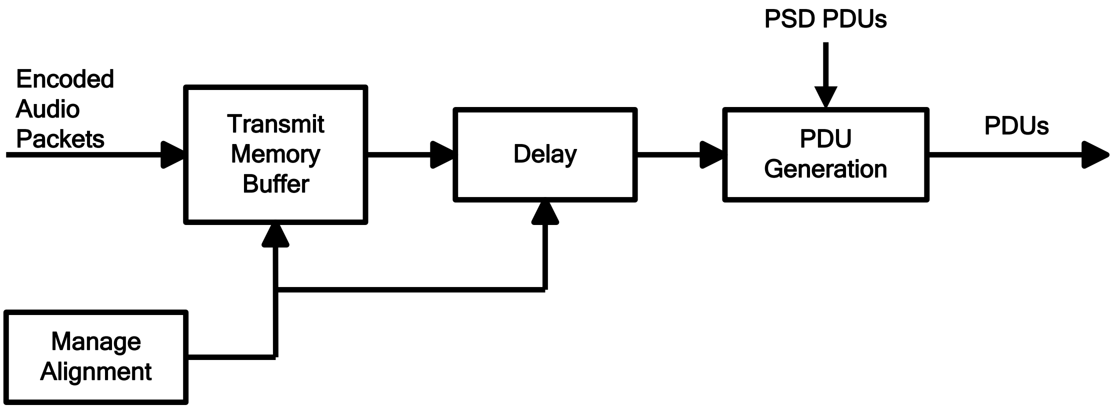


Figure 4-4: Audio Transport Block Diagram

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.

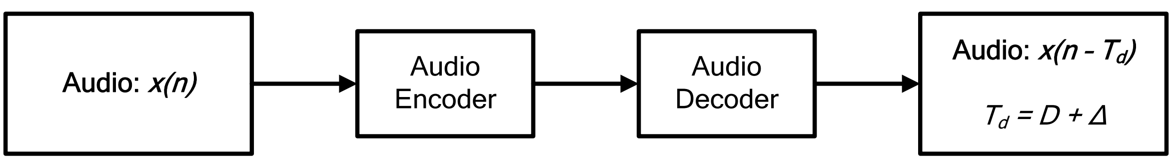


Figure 4-5: Audio Encoder/Decoder as a Constant Delay Element for the System

Another form of blending in the Hybrid AM and Hybrid FM systems applies to SPS. For SPS, no specific time alignment is required, as the blend involves transition between audio and silence.

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. Additional delay compensation may be requested in the units of audio samples, and the decoded audio stream is delayed by that amount.

Another case requiring proper handling of delay information on both the encoding side and the decoding side involves multi-stream programs. The enhanced stream must be aligned with the core stream; otherwise audio distortion may occur at the decoder output. In the case of multi-stream programs, the core stream is the base audio stream (where delay parameters are irrelevant) and only the enhanced stream carries relevant delay information.

4.4 Data Transport Interface

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

- Interface for injecting PSD into a 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 (although it is not necessarily used) for each defined audio stream. A PSD byte-stream may be added to each individual PDU in time-varying amounts. This is enabled through the PSD Transport [10]. 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 audio packet in each stream to other processes in the transmission system.

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 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 a 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 (that includes the RS parity bytes and the PDU Control Word), a variable number of audio packet location fields (Loc 0, Loc 1, …, Loc NOP-1), an optional Header Expansion field, PSD field, and encoded audio packets. The PDU Control Word is protected by a 96-byte Reed-Solomon (RS) codeword. Since the RS codeword is a fixed size it may also span portions of the Header Expansion field, PSD field, and possibly the encoded audio packets.

Each PDU consists of RS parity bytes, the PDU Control Word, the audio packet location fields (locators), optional Header Expansion, and a variable number of encoded audio packets. Each locator (Loc) points to the CRC byte that follows the packet that it covers, using one locator per packet. The size of the locator fields (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 PDUs). This scenario is necessary to ensure that the 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.

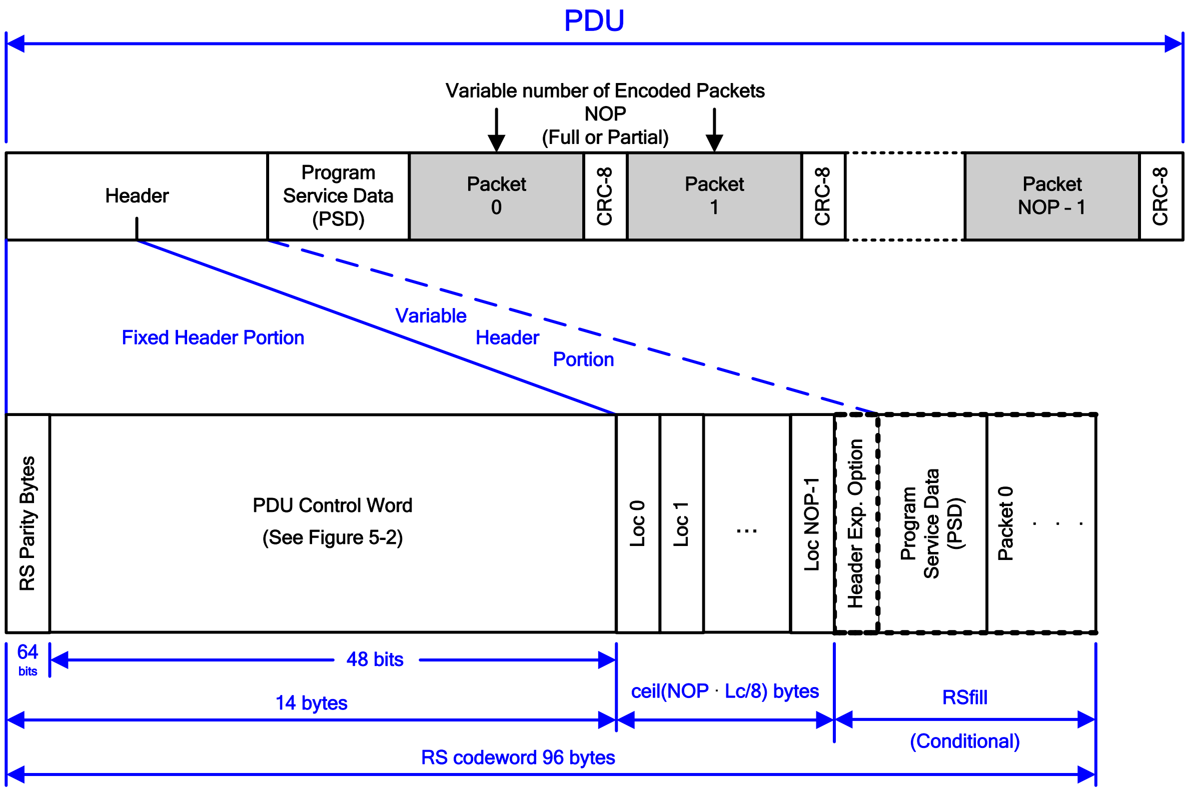


Figure 5-1: PDU Format

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, PDU control word, and audio packet locators may be less than the 96-byte RS codeword. When this case occurs, additional fields (Header Expansion, PSD, and Packet 0) fall within the 96-byte RS codeword and will be included in the RS parity byte computation.

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

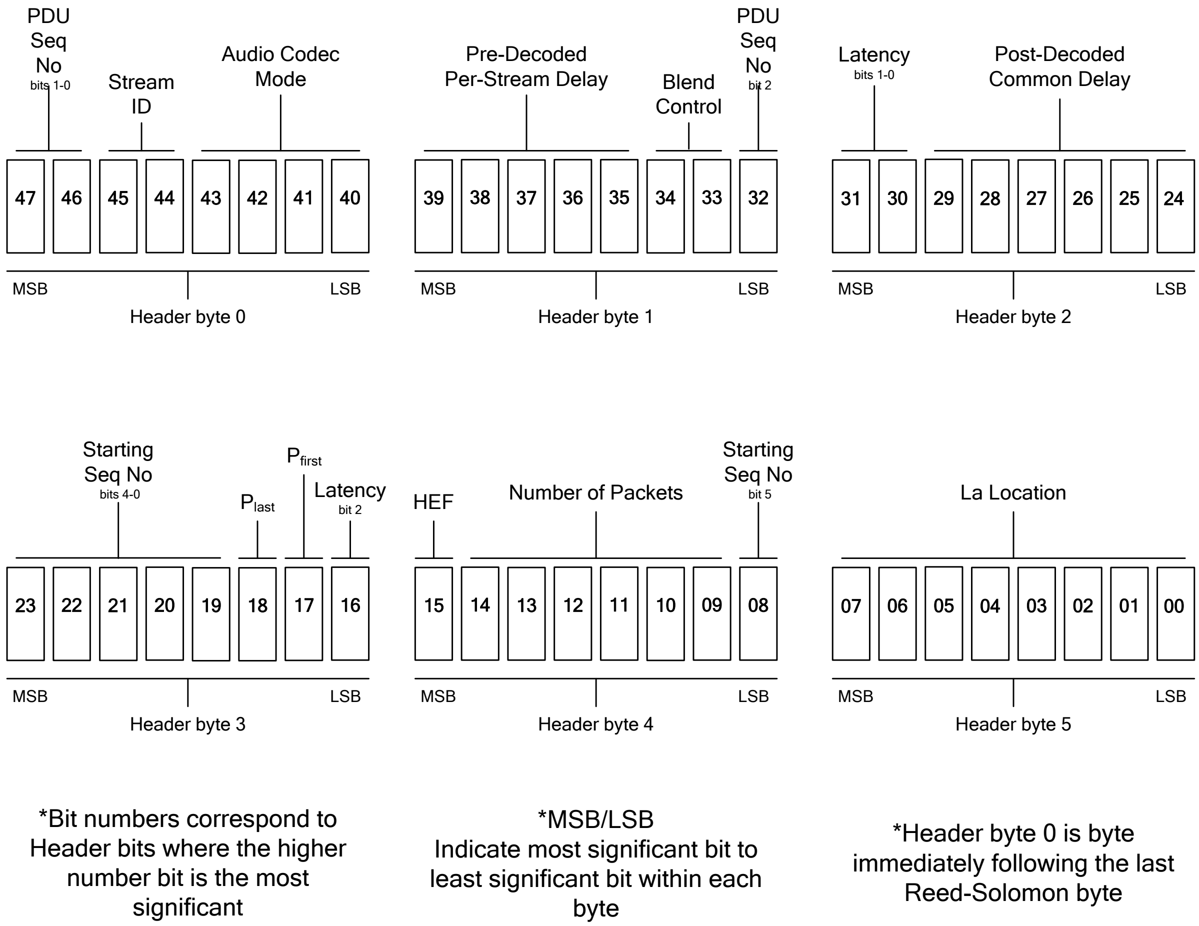


Figure 5-2: PDU Control Word – Bit Allocation for Codec Modes 0b0000 through 0b1110

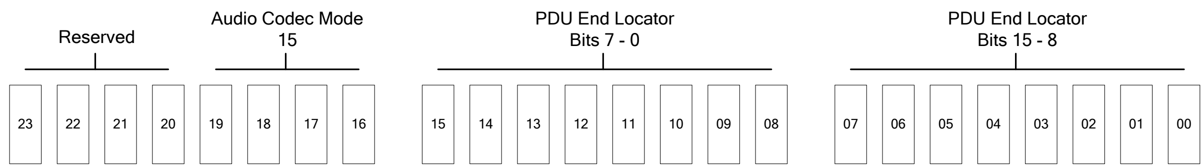


Figure 5-3: PDU Control Word – Bit Allocation for Codec Mode 0b1111 (15)

Details of the PDU format for Codec Modes 0b0000 (zero) through 0b1111 (15) are presented in Table 5-1 and the ensuing subsections.

Table 5-1: PDU Header Field Definitions

Bits	Field Name/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.

Bits	Field Name/Description	Comments
2	Stream ID	This field is used to indicate the stream type Core (0b00) Enhanced (0b01) Reserved (0b10) and (0b11)
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 5-4. For SPS PDUs, this field must be set to 0b00.
5	TX Digital Audio Gain Pre-decoded Per Stream Delay	If Stream ID = 0b00, this field defines the TX Digital Audio Gain (see Table 5-5). For all other Stream IDs, this field defines the delay between this corresponding stream and stream 0b00. Stream 0b00 is never delayed but the corresponding stream is delayed. This delay (in units of four-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 four-audio-frame periods) that the receiver uses to align the digital audio content with the 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 two-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 0b1 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 0b1 if the last encoded audio 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 0b1 when the optional Header Expansion Field is inserted immediately following the Location Fields.
8	La Location	Location of last byte of the Program Service Data field. The location is relative to the first byte (Byte 0) of the PDU.

Bits	Field Name/Description	Comments
NOP·Lc	Locator fields	Each of the 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 (Byte 0) of PDU.
0 - 128 (optional)	Header Expansion Field(s) (optional)	Header Expansion Field present only when Header Expansion Flag (HEF) is set to 0b1. See Table 5-6 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 encoded audio packets when the Expansion bytes and PSD do not extend to the RS codeword length (96 bytes).

The PDU header for codec mode 0b1111 (15) is shown in Figure 5-3. It includes the indication of the codec mode itself, followed by a 16-bit "PDU end locator" that points to the end of the PDU (i.e., the last byte of the PDU). The end of the PDU refers to the first byte of the PDU as byte number 0 (zero).

5.2.1.1 Audio Codec Mode

Table 5-2 defines the FM and AM audio codec modes according to stream configuration, PDU configuration, and bit rates. Table 5-2 shows the maximum bit rate for each codec mode in the context indicated by the "Typical Use" column. Note that higher bit rates may need to be allocated within the broadcast system in order to account for additional overhead, such as the audio header information.

The codec bit rate may be scaled back to a value less than the maximum rate. The minimum bit rate is dependent on the actual codec implementation and as well as the desired minimum acceptable audio quality. As technology improves, it may be possible that for a given level of audio quality, the bit rate can be reduced. Hence this document does not impose specific minimum bit rates for each codec mode. This is left as an implementation detail.

For two-stream audio codec modes, the broadcast system may further lower the configured bit rate by not using the enhanced stream.

Single-stream audio codec modes may contain a monaural (mono) or two-channel audio (stereo) signal. This also applies to two-stream systems when both streams are valid at the decoder input. The core audio stream of a two-stream signal that is valid at the decoder input may be either stereo or mono depending on the audio codec mode and the configured bit rate. A monophonic signal (L and R) is always decoded or presented as a dual-mono (that is, identical left and right channels) signal by the audio decoder.

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	Maximum Bit Rate (kbit/s)
0b0000	FM Hybrid	1	00	Core	1	32	0 - 1	16	96
0b0001	FM All Digital	2	00	Core	8	4	0 - 7	12	48
			01	Enhanced	1	32	0 - 1	16	48
0b0010	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
0b0011	FM All Digital	2	00	Core	8	4	0 - 7	12	24
			01	Enhanced	1	32	0 - 1	16	72
0b0100 to 0b1001	Reserved								
0b1010	FM Hybrid / All Digital	2	00	Core	1	32	0 - 1	12	22
			01	Enhanced	8	4	0 - 7	12	24

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	Maximum Bit Rate (kbit/s)
0b1011 to 0b1100	Reserved								
0b1101	FM Hybrid / All Digital	1	00	Core	8	4	0 - 7	12	24 / 96*
0b1110	Reserved								
0b1111	Reserved								

*For standard/advanced service modes, respectively.

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 5-2 shows 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 5-3 shows the stream compatibility for all the reserved codec modes.

Table 5-3: Backward Compatibility of Reserved Audio Codec Modes – Defaults

Audio Codec Mode	Default Audio Mode	Backward Compatible Streams	Stream Free to be Redefined†
0b0100	0b0010	0b00, 0b01	None
0b0101	0b0010	0b00, 0b01	None
0b0110	0b0010	0b00	0b01
0b0111	0b0010	0b00	0b01
0b1000	0b0000	0b00	None
0b1001	0b0011	0b00, 0b01	None
0b1011	0b0011	0b00	0b01
0b1100	0b0001	0b00, 0b01	None
0b1110	0b0001	0b00	0b01
0b1111	None	None	All

† Additional streams are assumed in expanded All Digital system service modes and possibly in advanced (future) hybrid configuration.

5.2.1.2 Blend Control

Table 5-4 defines the blend control bits. This definition applies only to the Main Program Service Audio (MPSA); for any Supplemental Program Service Audio (SPSA), the bits shall always be set to 0b00.

There are three blending states that are controlled by the value of the two blend control bits.

1. BLEND DISABLE (0b00): The broadcast system is configured in this state only when an all-digital waveform is transmitted. Analog diversity delay is not applied by the transmitter. The receiver shall disable analog blending.

2. BLEND SELECT (0b01): The broadcast system is configured in this state in one of two cases:
   - In hybrid and extended hybrid service modes with associated analog/digital audio content (MA1, MP1, MP2, MP3, MP11, MP1X, and MP1XOV), the broadcaster disables analog diversity delay and the receiver forces selection of analog audio. This state, also known as “ballgame mode,” allows receivers to output analog audio that is synchronous with live events.
   - In FM no-blend extended hybrid service modes (MP5, MP6, DSB1, DSB1OV, and MP6OV), the broadcaster disables analog diversity delay, and the receiver disables analog blending. The receiver allows the end user to manually select analog or digital audio.

     In FM no-blend extended hybrid service modes, the audio programs in the analog and digital portions of the extended hybrid waveform are not necessarily the same. Therefore, blending is not allowed, and analog diversity delay is not applied.

     Under no circumstances shall receivers automatically force digital audio in this state.

     This state is invalid when an all-digital waveform is transmitted.

3. BLEND ENABLE (0b10): The broadcast system is configured in this state only when a hybrid or extended hybrid waveform is transmitted. Analog diversity delay is applied by the transmitter and the receiver automatically blends between analog and digital audio, depending on the quality of the received digital signal.

   This state is invalid for all-digital and FM no-blend extended hybrid service modes.

A blend control setting of 0b11 is reserved for future use.

Table 5-4: Blend Control Bit Definitions

Audio Control Word Bit 34	Audio Control Word Bit 33	Waveform	Service Mode	Definition
0	0	AM and FM Hybrid	MA1 MP1	Not Valid 0b00 shall never be sent in the MPS PDUs in hybrid service modes. This can cause undefined behavior in various receiver models.
		FM Extended Hybrid	MP2 MP3 MP11	
		FM Extended Hybrid Advanced Service Modes (backward-compatible service modes)	MP1X MP1XOV	
		FM No-Blend Extended Hybrid (including advanced service modes)	MP5 MP6 / MP6OV DSB1 / DSB1OV	
		AM and FM All Digital	MP5 MP6 MA3	"BLEND DISABLE" state. 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 modes.
		FM All Digital Advanced Service Modes	DSB1 DSB1OV MP6OV	
0	1	AM and FM Hybrid and FM Extended Hybrid (including backward-compatible advanced service modes)	MA1 MP1 MP2 MP3 MP11 MP1X MP1XOV	"BLEND SELECT" state. No analog diversity delay has been applied by the transmitter. Analog and digital audio content is the same. RX shall disable analog blending and output analog audio for hybrid and extended hybrid waveforms in "ballgame mode".

Audio Control Word Bit 34	Audio Control Word Bit 33	Waveform	Service Mode	Definition
		FM No-Blend Extended Hybrid (including advanced service modes)	MP5 MP6 DSB1 DSB1OV MP6OV	“BLEND SELECT” state. No analog diversity delay has been applied by the transmitter. Analog and digital audio content may be different. For FM No-Blend Extended Hybrid Service Modes, the receiver may output analog or digital audio, as selected by the end user.
		AM and FM All Digital	MA3 MP5 MP6	Not Valid because no analog signal is transmitted. A Blend Control value of 0b01 shall not be transmitted for these cases.
		FM All Digital Advanced Service Modes	DSB1 DSB1OV MP6OV	
1	0	AM and FM Hybrid	MA1 MP1	“BLEND ENABLE” state. Analog diversity delay has been applied by the transmitter. RX shall automatically blend to analog when the digital audio quality measure is below the selected threshold.
		FM Extended Hybrid	MP2 MP3 MP11	
		FM Extended Hybrid Advanced Service modes (backward-compatible service modes)	MP1X MP1XOV	
		AM and FM All Digital	MA3 MP5 MP6	Not Valid because no analog signal is transmitted. A Blend Control value of 0b10 shall not be transmitted for these cases.
		FM All Digital Advanced Service Modes	DSB1 DSB1OV MP6OV	

Audio Control Word Bit 34	Audio Control Word Bit 33	Waveform	Service Mode	Definition
		FM No-Blend Extended Hybrid (including advanced service modes)	MP5 MP6 / MP6OV DSB1 / DSB1OV	Not Valid because analog diversity delay has not been applied by the transmitter. A Blend Control value of 0b10 shall not be transmitted for these cases.
1	1	All	MA1 MA3 MP1 MP2 MP3 MP11 MP5 MP6 MP1X MP1XOV DSB1 MP6OV DSB1OV	Reserved

5.2.1.3 TX Digital Audio Gain

For audio stream ID 0b00, header bits 39:35 provide the TX Digital Audio Gain parameter. Refer to Table 5-5 for a definition. For MPS audio, this field defines the audio level adjustment to be applied to the digital audio by the receiver in order to equalize the subjective loudness of the digital audio compared to the analog audio. For SPS audio, this field defines the audio level adjustment to be applied to the digital audio by the receiver in order to equalize the subjective loudness of the digital audio of the current program compared to that of the other audio programs.

Table 5-5: TX Digital Audio Gain Control

Value	Indicated RX Digital Audio Level Adjustment
Reserved	…
0b11000	-8 dB
0b11001	-7 dB
0b11010	-6 dB
0b11011	-5 dB
0b11100	-4 dB
0b11101	-3 dB
0b11110	-2 dB
0b11111	-1 dB
0b00000	0 dB
0b00001	+1 dB
0b00010	+2 dB
0b00011	+3 dB
0b00100	+4 dB
0b00101	+5 dB
0b00110	+6 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 PDU header. The HEF is set to 0b1 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 Header Expansion Fields.

5.2.1.5 Locator Fields

The bit organization of 16-bit and 12-bit locator fields is shown in Figure 5-4 and Figure 5-5 respectively. Each of the NOP locators points to the location of the last byte of the corresponding encoded audio packet for each of the NOP packets (partial or full) in the PDU. The location is relative to the first byte of the PDU (Byte 0). Each 16-bit locator consists of two 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.

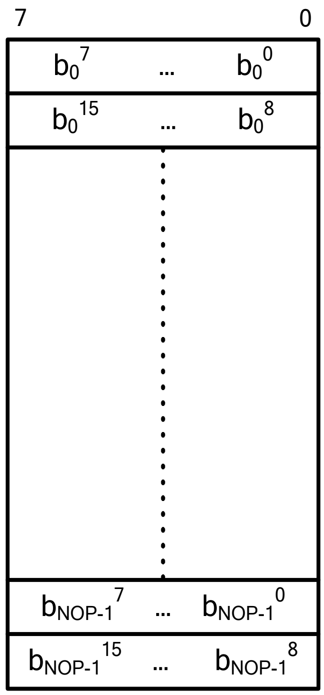
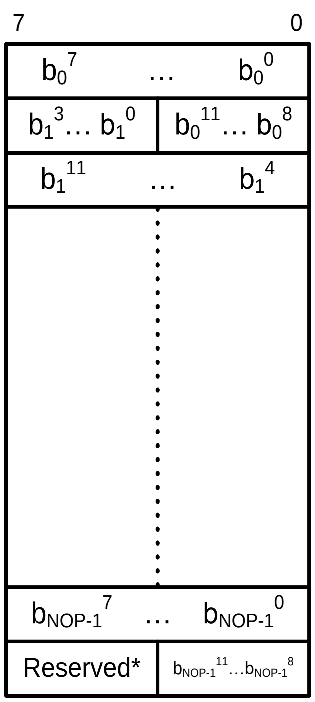


Figure 5-4: Locator Fields – 16-Bit



* is unused when NOP is odd

Figure 5-5: Locator Fields – 12-Bit

5.2.1.6 Header Expansion Fields

The header expansion field bits are defined in Table 5-6.

Table 5-6: Header Expansion Field Bits – Definitions

<table>
<tr><th># of 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 (HEF) is set to 0b1 when optional additional Header Expansion fields are inserted immediately following the HEF</td></tr>
<tr><td rowspan="9">3</td><td rowspan="9">Header Expansion ID</td><td>Bits</td><td>Header Expansion Field Indications/Formats/Types</td></tr>
<tr><td>0b000</td><td>Class Indication

This value for the Header Expansion ID field indicates the Class and is provided according to the description in Subsection 5.2.1.6.1</td></tr>
<tr><td>0b001</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 5.2.1.6.2</td></tr>
<tr><td>0b010</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 5.2.1.6.3</td></tr>
<tr><td>0b011</td><td>Reserved (Pre-defined)
For details see Subsection 5.2.1.6.4</td></tr>
<tr><td>0b100</td><td>PDU Marker Indication

This value for the Header Expansion ID field indicates that the PDU Marker is provided according to the description in Subsection 5.2.1.6.5</td></tr>
<tr><td>0b101</td><td>Reserved</td></tr>
<tr><td>0b110</td><td>Reserved</td></tr>
<tr><td>0b111</td><td>Reserved</td></tr>
<tr><td>N/A</td><td>Expansion Content</td><td colspan="2">Header Expansion Content – depends on the Header Expansion ID</td></tr>
</table>

As shown in Table 5-6, 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 expansion fields. It is set to 0b1 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 one byte. Each HEF indication is associated with the next header expansion byte immediately following. The HEF bit in the last Header Expansion field is set to 0b0.

The maximum number of Header Expansion field bytes is limited to 16. In streams containing PSD, the expansion bytes will reduce the number of PSD bytes that can be sent for that PDU. If there is no PSD present or if the expansion bytes exceed the allocated PSD capacity, then the system has the option to dynamically scale back the number of bytes allocated for audio to allow for header expansion bytes. The location of the last byte of PSD – La Location – points to the last location of the Expansion Fields when it exceeds the amount of PSD.

Figure 5-6 shows the Header Expansion Flag and the indication of additional Header Expansion Fields.

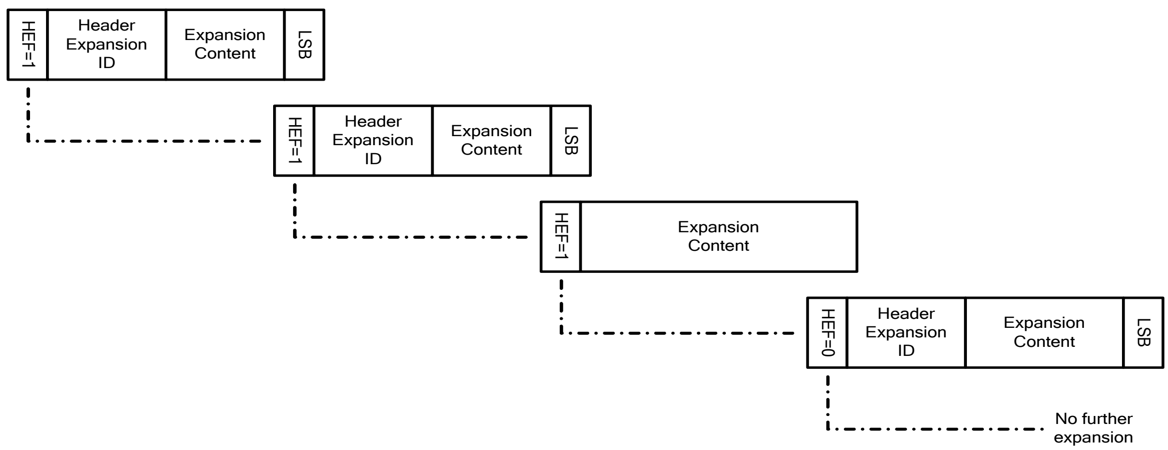


Figure 5-6: Header Expansion Fields – Example

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 5-6, the format of the Expansion Content field depends on the Header Expansion ID and may contain any or all of the following indications:

a) Class Indication – To identify and manage different classes of programs

b) Program Number – To identify and manage programs

c) Program Type – To identify the different program types and access type to the program

d) PDU Marker – A sequential number indicating the PDU placement in a sequence of PDUs

The rules for including a header expansion indication are as follows,

i. If a single PDU contains more than one Header Expansion Field, then the fields shall be arranged in an ascending order based on the Header Expansion ID.

ii. Any header expansion values not present in a PDU shall be considered 0 (zero) by the receiver audio transport. The exception to this rule is the class indication. If the class indication is not present in a PDU, then class does not apply.

iii. MPS does not have to include Program Number, unless PDU displacement (see displacement bit in Figure 5-9) is indicated or additional header expansions are included.

iv. SPS must include Program Number. Other fields are optional, unless they carry information that is necessary for proper use of the audio program (or audio segment in the program).

v. When a program has access limitations (Conditional Access), Class Indication (Header Expansion ID 0b000), Program Number Indication (Header Expansion ID 0b001) and Program Type Indication (Header Expansion ID 0b010) must be included.

5.2.1.6.1 Class Indication

If the Header Expansion ID is set to 0b000 (Class Indication), then the Expansion Content indicates the Class of the current program.

The Class Indications are listed and described in Table 5-7.

Table 5-7: Class Indications

Value	**Indicated Class**	**Comments**
0b0000	Restricted	0b0000 shall not be used.
0b0001	CA Program	Must be present to indicate a program that is conditionally-accessed (Does not apply to program number 0).
0b0010 to 0b1111	Reserved	---

Class Indication, when present, shall always be the first header expansion in the PDU. Class Indication may be used to provide additional information in association with the Program Number.

Class 0b0000 is restricted and cannot be defined for future applications. Class 0b0001 indicates a conditionally-accessed program. For MPS, Class Indication 0b0001 shall not be used since MPS is defined as a free-access program. For free-access SPS programs, class 0b0001 shall not be sent.

If class 0b0001 is sent, then the Access bit in header expansion 0b010 (Program Type Indication) shall also be sent with the Access bit set to 0b1.

Note that a program may be indicated as being conditionally accessed, but may or may not be scrambled (encrypted). For example, a program may normally be conditionally accessed, but may contain unencrypted material for some period of time that is of interest to all listeners.

5.2.1.6.1.1 Header Expansion for Class Indication

The Header Expansion for the Class Indication is shown in Figure 5-7.

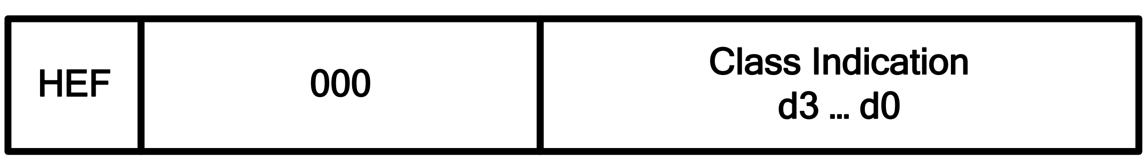


Figure 5-7: Header Expansion – Class Indication

5.2.1.6.2 Program Number Indication

If the Header Expansion ID is set to 0b001 (Program Number indication), then the Expansion Content contains the Program Number for the current PDU. The program number, together with the Class Indication uniquely identifies the PDU content.

The Program Number indication is used by the Audio Transport to identify and manage the MPS or SPS programs that are transmitted. For the Main Program Service, the program number shall be set to 0b000 and the Class Indication shall not be sent. For SPS, where the Class Indication is either set to 0b0001 (conditionally accessed) or is not sent (not conditionally accessed), the program number can be designated by the broadcasting system using any non-zero Program Number (0b001 through 0b111). For SPS, where the Class Indication is sent and is a number other than 0b0001, the Program Number can be designated by the broadcasting system using any number (0b000 through 0b111). SPS programs are not required to be numbered sequentially.

A supplemental program can be added or removed without affecting the enumeration of the main program or existing supplemental programs being transmitted. Unless the Class Indication is used, the Program Number shall always be the Header Expansion that occurs first in the PDU. It must be placed before the Program Type, Program Segment ID, or PDU Marker. For MPS, the Program Number Indication is sent only if additional Header Expansion fields are used to indicate Program Type, Program Segment ID, or PDU Marker. For SPS, the Program Number Indication is always sent.

5.2.1.6.2.1 Header Expansion for Program Number

The Header Expansion for the Program Number indication is shown in Figure 5-8.

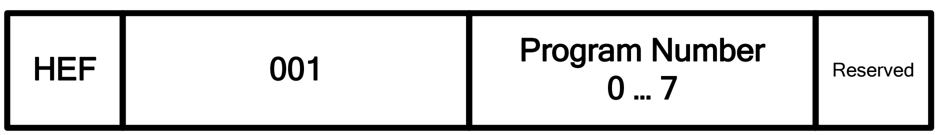


Figure 5-8: Header Expansion – Program Number Indication

The Expansion Content contains the Program Number indication (0 to 7). The LSB is indicated as "Reserved". By default, it has to be set to 0b0. However, setting it to 0b1 indicates that the Program Number indication is followed by two additional bytes of reserved header expansion content, as shown in Figure 5-9.

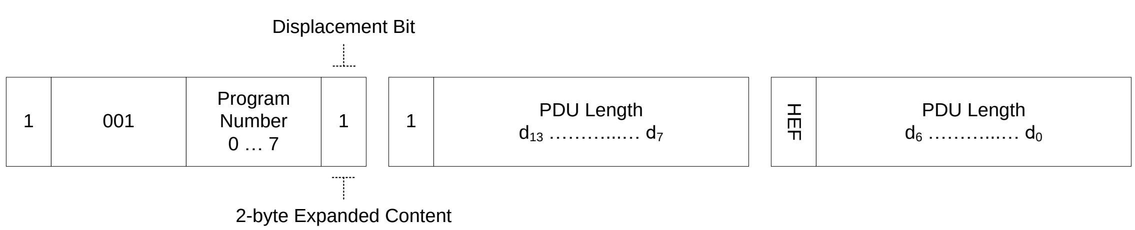


Figure 5-9: Header Expansion – 2-byte Expansion

If the displacement bit is set, two additional expansion bytes are used to indicate the displacement. The displacement points to the last byte of the PDU. Any data placed between the last audio packet locator and the indicated end of the PDU is reserved for future use.

5.2.1.6.3 Program Type Indication

If the Header Expansion ID is set to 0b010 (Program Type Indication), then the Expansion Content contains the Program Type. The Program Type Indication represents the program content by classifying it as one of the pre-defined types that are familiar to the receiver. It enables the receiver to search and sort through the variety of program content being broadcast.

The audio program types numbered zero through 31 are defined for use in the HD Radio system and are in accordance with NRSC-4 (Reference [29]). Additional program types can be included as required, according to https://hdradio.com/broadcasters/us-regulatory/nrsc-supplemental-information/.

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

The Program Type Header Expansion includes a 1-bit flag to indicate the designated accessibility to the program. When this flag is set to 0b0 (zero), it indicates that the program is free of any access limitations. When this flag is set to 0b1 (one), it indicates that the program is delivered with access limitations (Conditional Access). The actual details of the instantaneous access conditions are conveyed separately, i.e., a program may be designated as a CA program but may or may not be encrypted.

When no access limitations are applied, the Program Type need not be transmitted. When access limitations apply, the Program Type shall be transmitted. It is transmitted every PDU while in audio codec modes 0b0000 or 0b1010. It is transmitted at least once every eight PDUs while in audio codec modes 0b0001, 0b0010, 0b0011 or 0b1101.

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 [6] for details.

The Program Type Expansion Content in the Program Type Header Expansion can be changed without any interruption to the current program.

5.2.1.6.3.1 Header Expansion for Program Type

The Header Expansion for the Program Type indication is shown in Figure 5-10.

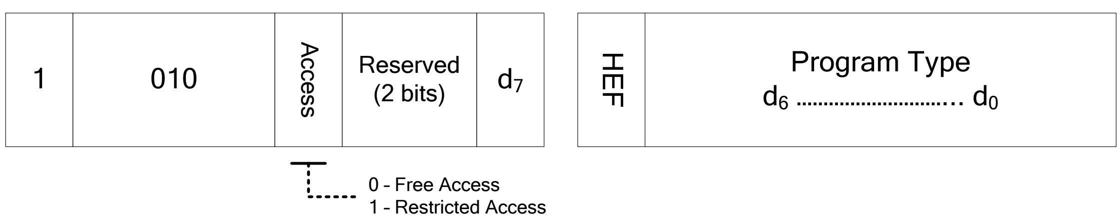


Figure 5-10: Header Expansion – Program Type

The Header Expansion for the Program Type indication consists of two expansion bytes. The first byte contains the Header Expansion ID for the Program Type. It also contains the “Access” flag. The Program Type number is then constructed in the second expansion byte which contains the seven LSBs of this Program Type number. The LSB of the first byte is then used as the MSB of the second byte to construct the actual 8-bit Program Type number. There are two bits of Reserved Expansion Content in the first byte and these two bits may be used to further expand the Program Types.

5.2.1.6.4 Reserved Header Expansion

If the Header Expansion ID is set to 0b011 then the Header Expansion is pre-defined as shown in Figure 5-11.

5.2.1.6.4.1 Header Expansion for Program Segment ID

The Header Expansion for the Program Segment ID is shown in Figure 5-11.

Figure 5-11: Header Expansion – Program Segment ID

5.2.1.6.5 PDU Marker Indication

If the Header Expansion ID is set to 0b100 (PDU Marker indication), then the Expansion Content contains the PDU Marker.

The HD Radio broadcasting system generates and applies the PDU Marker in the Header Expansion fields. It is an optional parameter that is used to uniquely identify the PDU in a program, for a purpose, such as access control management to programs. The default start value is zero.

For a multi-stream configuration, the PDU Marker is indicated only on the core stream.

When no access limitations are applied or no reference is made to specific audio PDUs, the PDU Marker need not be transmitted. When access limitations apply or reference, for any specific purpose, is made to audio PDUs, the PDU Marker shall be transmitted. It is transmitted every PDU, while in audio codec modes 0b0000 or 0b1010. It is transmitted once every eight PDUs, within PDU Sequence Number 0 (bits 32, 46-47 in the PDU header), while in audio codec modes 0b0001, 0b0010, 0b0011 or 0b1101.

The Header Expansion for the PDU Marker indication can be changed without any interruption to the current program. The PDU Marker start number default is zero, but the broadcasting system may assign it as convenient. The number is incremented every long PDU (audio codec modes 0b0000 or 0b1010) and every short PDU coincident with L1 block number 0 (audio codec modes 0b0001, 0b0010, 0b0011 or 0b1101).

5.2.1.6.5.1 Header Expansion for PDU Marker

The Header Expansion for the PDU Marker indication is shown in Figure 5-12.

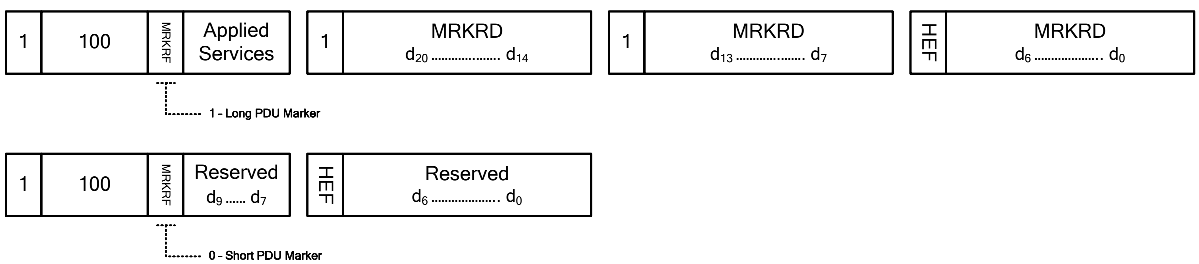


Figure 5-12: Header Expansion – PDU Marker

The first byte of the Header Expansion for PDU Marker contains a 1-bit field (MRKRF) to indicate the length of the PDU Marker header expansion: 0b1 indicates a 4-byte header expansion for PDU Marker and 0b0 indicates a 2-byte header expansion for PDU Marker. The 2-byte header expansion is reserved for future use as shown in Figure 5-12.

The first byte of a 4-byte Header Expansion for PDU Marker ("Long PDU Marker") contains a 3-bit field that indicates the "Applied Services" in the PDU payload. This is further explained in Table 5-8. If the value for the "Applied Services" indication is set to 0b001, then this indication signals that the audio is encrypted. When no processing / encryption is applied (conveyed/signaled by a value of 0b000 for the "Applied Services" indication), the header expansion still indicates the PDU Marker. The Applied Services Indication is allowed with every increment to the marker. Thus, content encryption may be dynamic, and may change on any marker boundary.

If the value for the "Applied Services" indication is set to 0b001 and a receiver is not entitled for the encrypted service, the receiver shall inhibit the PSD data and the audio PDU will not be sent to the audio decoder within the receiver.

The PDU Marker consists of 21-bits which span across the remaining three bytes of the Header Expansion. The PDU Marker resolution is a single long PDU (L1 frame equivalent) for all audio codec modes. For audio codec modes 0b0000 or 0b1010, it is also indicated at a long PDU rate. For audio codec modes 0b0001, 0b0010, 0b0011, or 0b1101, which employ short PDU, it can be indicated at a short PDU rate, but the indicated marker value can only change at a long PDU rate. This means that the PDU Marker resolution is equivalent to one L1 frame period, and thus also equivalent to an ALFN unit.

The PDU Marker is continuous, which means that its same-number representation repeats every 2^{21} L1 frames. Thus, the PDU Marker spans approximately 36 days.

Table 5-8: Applied Services Indication

Applied Services Indication	Applied Services Meaning
0b000	None – Content is not altered
0b001	Encoded audio packets are encrypted. This Indication shall only be used if: Class Indication (Expansion byte 0b000) = 0b0001 and Access bit of Expansion Byte 0b010 = 0b1 to indicate a CA service
0b010 to 0b111	Reserved

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 [10]. A PSD byte-stream is supported by each active encoder stream but utilized for the core stream only. Refer to Table 5-9 for the typical/average number of data bytes per PDU per stream for each of the various audio codec modes. Rates are not guaranteed, as they may be instantaneously reduced by Header Expansion. The rates may also be instantaneously increased, as a result of the audio format. They may also be affected by different configurations, especially at low audio bit rates.

Table 5-9: Program Service Data per Audio Codec Mode

Audio Codec Mode	Typical Use	Stream ID (bits)	Stream Type	Nominal Program Service Data Rate (Bytes/PDU)
0b0000	FM Hybrid	00	Core	128
0b0001	FM All Digital	00	Core	16 §
		01	Enhanced	0
0b0010	AM Hybrid / All Digital	00	Core	16 §
		10	Enhanced	0
0b0011	FM All Digital	00	Core	16 §
		01	Enhanced	0
0b0100 to 0b1001	Reserved	-	-	-
0b1010	FM Hybrid / All Digital	00	Core	128
		10	Enhanced	0
0b1011 to 0b1100	Reserved	-	-	-
0b1101	FM Hybrid / All Digital	00	Core	16 §
0b1110 to 0b1111	Reserved	-	-	-

§ Nominally, 16 bytes are allocated for PSD. The actual number of bytes for PSD must be decreased by the number of header expansion bytes utilized within a given PDU. In addition, the actual number of available bytes for PSD may be increased to up to 21 in a specific PDU, when not all of the bytes in that PDU are consumed by encoded audio packets.

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 Header 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 eight 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, where the LSB is on the right)

- 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 audio 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. Note that if a packet is split into two partial packets in a PDU, both partial packets will have their own individual CRC applied.

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 encoded audio packet.

5.2.4 Audio Encryption

When audio encryption is applied, encoded audio packets are encrypted. Each packet is encrypted separately and no byte aggregation or buffering across packets is allowed. For each packet, the first 64 bytes of the packet are encrypted. If the packet is shorter than 64 bytes, then the entire packet is encrypted. The packet encryption is not applied to the CRC-8 byte that is appended to each packet.

5.2.4.1 Marker Timing

For the purpose of synchronization between the audio PDU and the encryption data (codeword), the indicated PDU Marker applies at very specific times.

For Long PDUs, the marker is conveyed within every PDU and applies to that PDU.

For Short PDUs, the marker is conveyed within PDU number 0, and may also be conveyed within PDU number 3, but it applies from PDU number 0 to PDU number 7.

Since the PDU Marker resolution is one long PDU (one L1 frame), the encryption codeword may only change from one long audio PDU to another, but not faster.

5.2.4.2 Encoded Audio Packet Encryption

Encoded Audio Packet encryption is based on packet boundaries, rather than precise PDU boundaries. The encryption data (codeword) is applied to the first complete packet in the indicated PDU to the end of the last partial packet of the indicated PDU, even if it is partial and spills over to a newly indicated PDU.

Specifically, in long PDUs (core audio in audio codec modes 0b0000 or 0b1010), the same encryption data must be used from the first complete encoded audio packet of a PDU to the end of first partial packet in the next PDU.

In short PDUs (core audio in audio codec modes 0b0001, 0b0010, 0b0011 or 0b1101), the same encryption data must be used from the first complete encoded audio packet of a PDU number 0 through PDU number 0 to 7 and to the end of first partial packet in the next PDU number 0. This approach guarantees that the encryption time used by specific encryption data is the same, regardless of the specific audio transport (audio codec mode) used for the specific program.

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/SPS PDU	Refers to the output of the Audio Transport process in the broadcasting system. An MPS/SPS PDU consists of protocol information followed by a sequence of encoded audio packets. MPS/SPS 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.

HD Radio™ Air Interface Design Description Station Information Service Transport

Version K

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TRADEMARKS

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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 how control and information are passed through the SIS Transport for subsequent processing by Layer 2.

2 Referenced Documents

STATEMENT
Each referenced document that is mentioned in this document shall be listed in the following iBiquity document:

- Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard Document Number: SY_REF_2690s

3 Abbreviations and Conventions

3.1 Abbreviations and Acronyms

ADV	Advanced Processing
ALFN	Absolute L1 Frame Number
AM	Amplitude Modulation
ASCII	American Standard Code for Information Interchange
CAP	Common Alerting Protocol
CRC	Cyclic Redundancy Check
DST	Daylight Saving Time
EA	Emergency Alerts
EBU	European Broadcasting Union
FCC	Federal Communications Commission
FEMA	Federal Emergency Management Agency
FM	Frequency Modulation
GPS	Global Positioning System
IBOC	In-Band On-Channel
ID	Identification
ID3	Tag Embedded In MPEG I Layer III Files
ISO	International Organization for Standardization
L1	Layer 1
LLDS	Low-Latency Data Service
LSB	Least Significant Bit
MF	Medium Frequency
MIME	Multipurpose Internet Mail Extensions
MSB	Most Significant Bit
MSG	Message
PDU	Protocol Data Unit
PIDS	Primary IBOC Data Service Logical Channel
PIDSOV	Primary IBOC Data Service Overlay Logical Channel
RDS	Radio Data System
SIDS	Secondary IBOC Data Service Logical Channel
SIS	Station Information Service
UTC	Coordinated Universal Time
VHF	Very High Frequency
WGS	World Geodetic System

3.2 Presentation Conventions

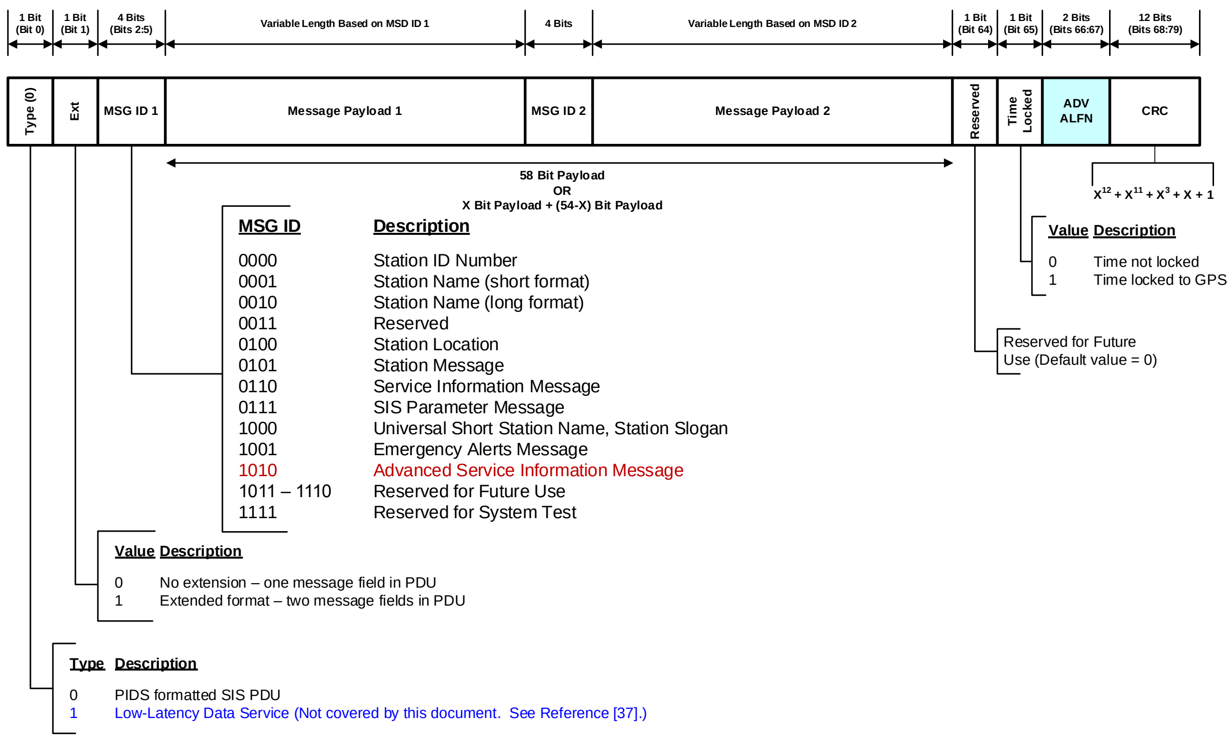
Unless otherwise noted, the following conventions apply to this document:

- 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 most 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 lowest index.
- In representations of binary numbers, the least significant bit is on the right.
- Hexadecimal numbers are represented by a prefix of “0x”

4 Station Information Service Protocol Data Unit Format

The Station Information Service (SIS) provides broadcast station identification and control information. SIS is transmitted in a series of SIS Protocol Data Units (PDUs) on the Primary IBOC Data Service (PIDS) logical channel. For more information on PIDS see [1] and [2]. SIS PDUs are 80 bits in length as shown in Figure 4-1. The most significant bit of each field is shown on the left. Layer 2 and Layer 1 process MSBs first; that is, bit 0 is the first bit interleaved by L1. The PDU contents are defined by several control fields within the PDU.

The Type bit is set to 0 for SIS PDUs. When the Type bit is 1, a low-latency data service (LLDS) PDU is sent instead. Since SIS shares the PIDS, PIDSOV, and SIDS logical channels with the LLDS transport, the high-level 80-bit PDU format is common to that of LLDS. This commonality simplifies multiplexing PIDS, PIDSOV, and SIDS capacity between the two services. LLDS is supported by only a subset of advanced FM service modes and by none of the standard service modes. Refer to [37] for more information on LLDS.



NOTES

MSG ID [0010] Station Name (long format): Not recommended for new designs. See Subsection 4.2.2 for details.

MSG ID [0011] ALFN: Discontinued; MSG ID [0011] is not needed since ALFN is already sent serially in the "ADV ALFN" field in the SIS PDU

Figure 4-1: SIS PDU Format – Type = 0

Type 0 PDUs may contain two, independent, variable-length, short message fields or a single longer message, depending on the state of the Ext bit. If the Ext bit equals 0, the message payload 1 field is up to

58 bits in length and the message contents are determined by the state of the first message ID 1 field. Any unused bits at the end of the message payload 1 field are set to zero. If the Ext bit equals 1, then the first message payload and contents are defined by MSG ID 1, and a second message may be present, with payload length and contents defined by MSG ID 2. In this case, the combined payload lengths of the two messages must be no greater than 54 bits. Any unused bits at the end of message payload 2 are set to zero.

The definitions of the MSG ID 1 and MSG ID 2 fields are identical. Refer to Table 4-1 for details of the MSG ID field. Any message may be placed in either message 1 or message 2 provided that the total available payload length is not violated. Longer messages must use the single message option (Ext = 0).

Table 4-1: MSG ID Definitions

MSG ID	Payload Size (bits)	Description	Comments
0000	32	Station ID Number	Used for networking applications Consists of Country Code and FCC Facility ID.
0001	22	Station Name – short format	Identifies the 4-alpha-character station call sign plus an optional extension
0010	58	Station Name – long format	Identifies the station call sign or other identifying information in the long format May consist of up to 56 alphanumeric characters Not recommended for new designs Must have message content that is identical to Station Slogan See Subsection 4.2.2 for details
0011	32	Reserved	Reserved
0100	27	Station Location	Provides the 3-dimensional geographic station location Used for receiver position determination
0101	58	Station Message	Allows a station to send an arbitrary text message
0110	27	Service Information Message	Identifies Program category of the Main and Supplemental programs. Introduces the data services
0111	22	SIS Parameter Message	Carries supplementary information, including Leap Second/Time Offset and Local Time data parameters
1000	58	Universal Short Station Name Station Slogan	Allows transmitting the station names up to twelve characters in length and supports international character sets
1001	58	Emergency Alerts Message	Allows for the provision of Emergency Alerts and follow-up information Allows for the “waking up” of a receiver

MSG ID	Payload Size (bits)	Description	Comments
1010	27	Advanced Service Information Message	Applicable to the P4 logical channel of MP1X and MP1XOV only. Identifies Program category of the Main and Supplemental programs. Introduces the data services.
1011 - 1110	TBD	Reserved	Reserved for future use
1111	TBD	Reserved	Reserved

The following subsections describe each message type (MSG ID).

4.1 Station ID Number (MSG ID = 0000)

This message type is uniquely assigned to each broadcasting facility. Figure 4-2 shows the message structure for the Station ID Number.

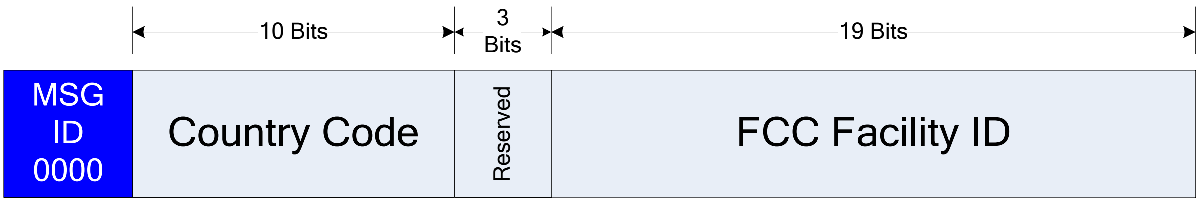


Figure 4-2: Station ID Number – Message Structure

Table 4-2 lists and describes the fields in the Station ID Number.

Table 4-2: Station ID Number – Field Names and Field Descriptions

Field Name	Number of Bits	Field Description
Country Code	10	In binary representation, the ten bits shall be used to represent the two-character country code as specified in Reference [14]
Reserved	3	Reserved bits default to “0”
FCC Facility ID (U.S. only)	19	Binary representation of unique facility ID assigned by the FCC in the U.S. Reference: [17]

Note that, although the FCC has expanded the width of its Facility ID to 20 bits, the HD Radio system will continue to broadcast only the 19 least significant bits, to maintain backward compatibility with legacy receivers.

With regard to the Country Code, the ISO 3166-1-alpha-2 code elements are two-letter codes. At the time of this publication, there were 249 code elements (that is, 249 countries) represented.

Table 4-3 maps each five-bit binary sequence to its decimal equivalent and its alpha character.

Table 4-3: Mapping Five-Bit Binary Sequences to Decimal Equivalents and Alpha Characters

Five-Bit Binary Sequence	Decimal Equivalent	Alpha Character
00000	0	A
00001	1	B
00010	2	C
00011	3	D
00100	4	E
00101	5	F
00110	6	G
00111	7	H
01000	8	I
01001	9	J
01010	10	K
01011	11	L
01100	12	M
01101	13	N

Five-Bit Binary Sequence	Decimal Equivalent	Alpha Character
01110	14	O
01111	15	P
10000	16	Q
10001	17	R
10010	18	S
10011	19	T
10100	20	U
10101	21	V
10110	22	W
10111	23	X
11000	24	Y
11001	25	Z

Note that the alpha characters are capital letters from the English alphabet.

As an example, using the details from Table 4-3, for the United States (US), the individual-letter decimal equivalents for the US would be 20 (U) and 18 (S); the individual-letter binary equivalents for the US would be 10100 (U) and 10010 (S).

To form a country code, these two five-bit binary numbers are concatenated to form a single 10-bit binary number. The left-most character is contained in the most significant bits. In binary, the country code (US) is 1010010010 and in decimal, the country code (US) is 658.

Other country code examples include:

Canada (CA) which is 0001000000 in binary and 64 in decimal,

Brazil (BR) is 0000110001 in binary and 49 in decimal, and

Mexico (MX) is 0110010111 in binary and 407 in decimal.

India (IN) is 0100001101 in binary and 269 in decimal.

4.2 Station Name

This message type has both a short format and a long format. The short format may be used with the two-message PDU structure so that it may be multiplexed with other messages and thus can be repeated frequently. The long format requires the single-message structure and may be extended across multiple PDUs. This format can be used to identify stations by a moderately long text string.

4.2.1 Station Name – short format (MSG ID = 0001)

Four-character station names may be broadcast with the short format. The field is 22 bits in length with the first bit on the left. Figure 4-3 shows the message structure for the Station Name (short format).

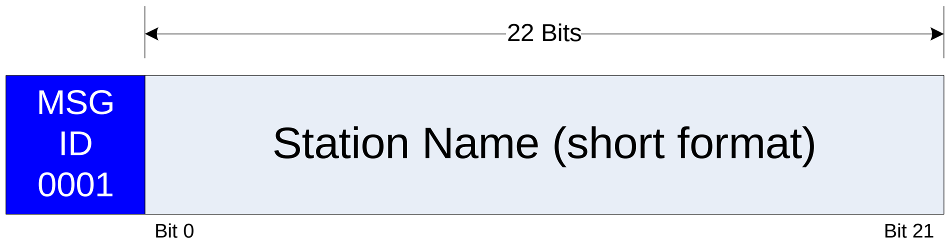


Figure 4-3: Station Name (short format) – Message Structure

Each character is five bits in length (MSB first, or leftmost), followed by a 2-bit extension. Refer to Table 4-4 for details of the field bit assignments (positions) and Table 4-5 for the character definitions. Only upper-case characters are defined, plus a limited number of special characters, as shown. The space character may be used, for example, to terminate a three-character call sign.

The first five bits are assumed to contain the leftmost character. For example, a station name of “ABCD” would be encoded in binary as 00000 00001 00010 00011 00. The 2-bit extension may be used to append an extension to the right of the other four characters (00 in the preceding example).

Table 4-4: Station Name (short format) – Field Bit Assignments (Positions)

Field Bit Positions	Description
0:4	Leftmost Character
5:9	Second Leftmost Character
10:14	Third Leftmost Character
15:19	Rightmost Character
20:21	Extensions: 00 = no extension 01 = Append “-FM” 10 = Reserved for future use 11 = Reserved for future use

Table 4-5: Station Name (short format) – Character Definitions

Value (MSB:LSB)	Character	
00000, 00001, 00010,…, 11001	A, B, C,…, Z	See Table 4-3
11010	space character	
11011	?	
11100	-	
11101	*	
11110	$	
11111	Reserved	

4.2.2 Station Name – long format (MSG ID = 0010)

This message type has been replaced with Station Slogan (MSG ID = 1000) and is no longer used for new applications; however, for countries where first-generation receivers are still in use, it is recommended that broadcasters continue to support this message type. In this case, both station slogan and long station name shall be broadcast and they must have identical message content.

The long format permits the station name to consist of textual strings. Each message contains seven characters encoded as 7-bit ISO-8859-1 characters [22]. In other words, it uses the first 128 characters of the ISO/IEC 8859-1 character set and the most significant bit of the 8-bit character code is truncated to form a 7-bit character.

The Station Name (long format) accommodates up to 56 characters in the name.

A character string may be extended over up to eight PDUs. The first three bits of the field specify the frame number of the last frame (or equivalently, the total number of SIS PDUs containing the message minus one) and the next three bits specify the frame number of the current PDU. PDU number zero is considered the leftmost of the string. The seven most significant station name bits within a PDU define the leftmost character for that PDU. For the last SIS PDU of the string, unused message bits are filled in with null characters (0x00).

The three LSBs of the Station Name message structure define the sequence number. This number is incremented modulo eight each time the character string is changed. The sequence number will only change within the PDU containing frame 0 of the message. All frames of the same message content will always have the same sequence number.

Figure 4-4 illustrates the message structure for Station Name (long format).

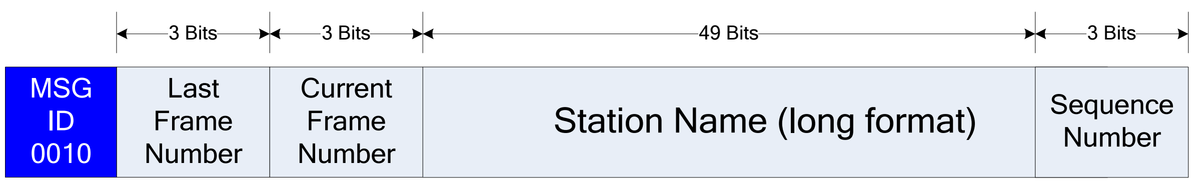


Figure 4-4: Station Name (long format) – Message Structure

As indicated in Figure 4-1 and Table 4-1, Station Name – long format (MSG ID = 0010) is not recommended for new designs.

4.3 (MSG ID = 0011) – Reserved

This message type is restricted and shall not be used for any purpose in the future.

4.4 Station Location (MSG ID 0100)

This message type indicates the absolute three-dimensional location of the feedpoint of the broadcast antenna. Such location information may be used by the receiver for position determination. The message structure is shown in Figure 4-5. Position information is split into two messages: a high portion and a low portion.

Altitude is in units of [meters·16] (that is, the LSB is equal to 16 meters). Altitude ranges between zero and 4080 meters. If the actual altitude is less than zero, it shall be set to zero (i.e., byte value of 0x00). If the actual altitude is greater than 4080 meters, it shall be set to 4080 meters (i.e., byte value of 0xFF).

Latitude and longitude are both in the same fractional formats. The LSB is equal to 1/8192 degrees. The MSB is the sign bit, which indicates the hemisphere. Positive latitude values represent positions north of the equator. Positive longitudes are in the eastern hemisphere. Longitude ranges are from -180 to +180, while permissible latitude values are between -90 and +90. Anything outside of these ranges is invalid. Refer to Subsection 5.1 for an example.

Used by the Global Positioning System, the World Geodetic System 84 (WGS 84) is used as the reference datum for location information. See Reference [18].

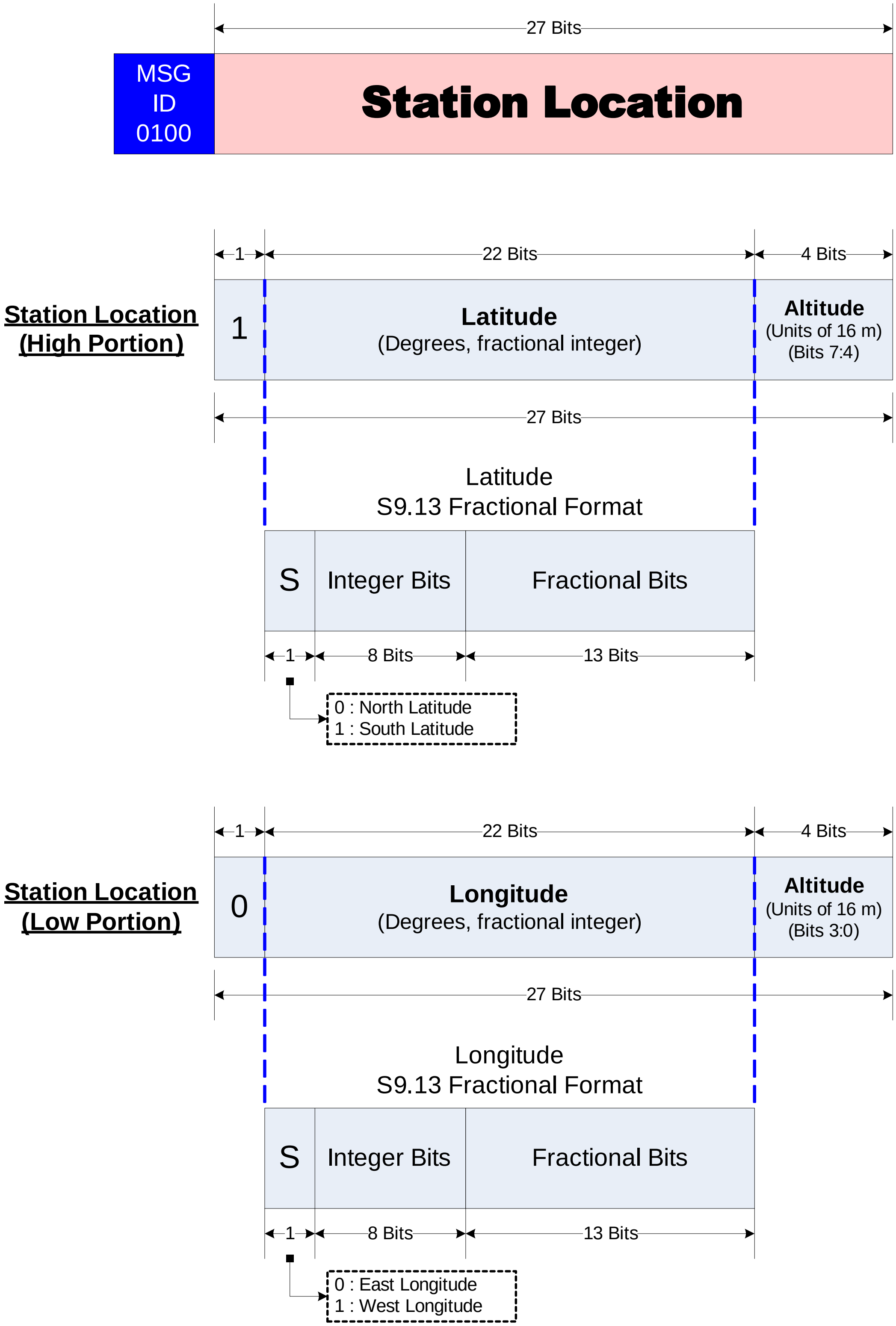


Figure 4-5: Station Location – Message Structure

4.5 Station Message (MSG ID 0101)

This message type allows the station to send any arbitrary text message. Examples include public service announcements, weather reports, or telephone call-in numbers. The Station Message has a total payload of 58 bits per frame. This message can span over multiple frames. Figure 4-6 shows the message structure for the Station Message. The format of the first frame is different from the others, as shown.

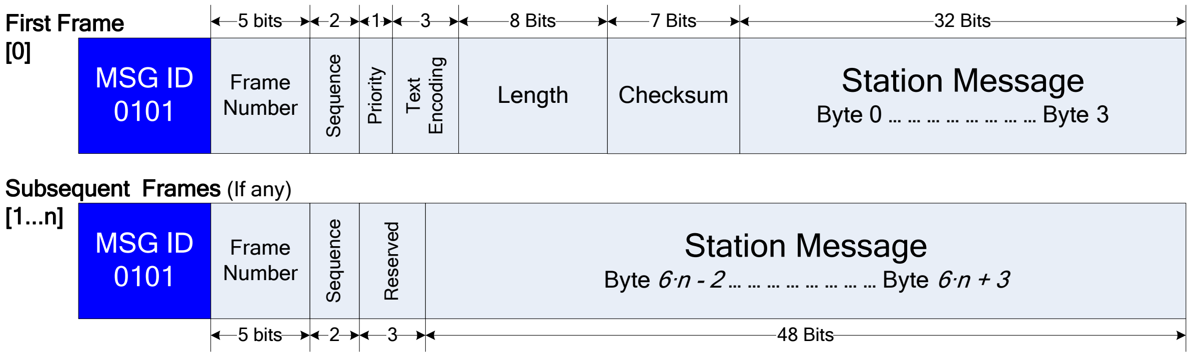


Figure 4-6: Station Message – Message Structure

The Station Message can be used to send a string of up to 190 8-bit characters [22] or 95 16-bit characters [23] per message. A message may span up to 32 frames. Each message contains a sequence number, indicating when the message text or priority has changed. A priority indicator is included to indicate that a message has an elevated importance. When multiple messages are broadcast, a message with the priority indicator set will advance to the top of the receiver queue. Any change in the message content or the priority is considered a new message and the sequence number is incremented. A 7-bit checksum is included in the first frame to increase receive reliability.

Table 4-6 and Table 4-7 describe the data fields for the first and subsequent frames, respectively.

Table 4-6: Description of Station Message Fields for Frame Number = 0

Field Name	Range	Description
Frame Number	0	Indicates the current frame number of the message Set to zero for the first frame
Sequence	0 - 3	Increments by 1, modulo 4, whenever the station message text and/or priority changes A new sequence number must commence with frame 0 and the same number shall be used for all frames of a given Station Message
Priority	0 - 1	Priority = 0: Normal priority Priority = 1: High priority When multiple Station Messages are broadcast, the receiver shall place a high priority message at the top of the queue as soon as it is received
Text Encoding	0 - 7	See Table 4-8
Length	4 - 190	Defines the total number of bytes of the Station Message text, excluding any unused bytes in the last frame For 16-bit character encoding, the Length must be even
Checksum	0 - 127	Checksum of all the data bytes of the Station Message text, excluding overhead bytes Refer to Figure 4-7 for details

Field Name	Range	Description
Station Message	N/A	For 8-bit character encoding, frame 0 contains the first 4 characters of the Station Message Byte 0 is the leftmost character For single-frame Station Messages, any unused bytes to the right of the Station Message text are filled with NULL characters (0x00) For 16-bit character encoding, frame 0 contains the first 2 characters of the Station Message Bytes 0:1 convey the leftmost character For single-frame Station Messages, any unused byte pairs to the right of the Station Message text are filled with NULL characters (0x00 00)

Table 4-7: Description of Station Message Fields for Frame Number = 1 to n

Field Name	Range	Description
Frame Number	1 - 31	Indicates the current frame number of the message
Sequence	0 - 3	Increments by 1, modulo 4, whenever the station message text and/or priority changes A new sequence number must commence with frame 0 and the same number shall be used for all frames of a given Station Message
Reserved	0 - 7	Reserved for future use
Station Message	N/A	For 8-bit character encoding, frames 1 to n contain the additional characters of the Station Message, where the lowest numbered byte within a frame is the leftmost for that frame For the last frame, any unused bytes to the right of the Station Message text are filled with NULL characters (0x00) For 16-bit character encoding, frames 1 to n contain additional characters of the Station Message, where the lowest numbered byte-pair within a frame is the leftmost for that frame For the last frame, any unused byte pairs to the right of the Station Message text are filled with NULL characters (0x00 00)

NRSC Supplemental Information provides the most up-to-date information regarding the Text Encoding Definitions: http://hdradio.com/broadcasters/us-regulatory/nrsc-supplemental-information.

Table 4-8: Text Encoding Definitions

Value	Field Definition
000 (default)	ISO/IEC 8859-1:1998 (Reference [22])
001 to 011	Reserved
100	ISO/IEC 10646-1:2000 UCS-2 (Little Endian) (Reference [23])
101 to 111	Reserved

Figure 4-7 illustrates the method used to calculate the 7-bit checksum. First, a 16-bit sum is computed by adding together all of the bytes of the station message text bytes (excluding overhead). The message bytes and the sum are both treated as unsigned integers. The 16-bit sum is then divided into a high (most significant) byte and a low (least significant) byte. The most significant bit of the high byte (bit 15 in Figure 4-7) is cleared. The high and low bytes are then summed together and the seven least significant bits of the sum are written into the checksum field, where the most significant bit is the left-most checksum bit shown in Figure 4-6.

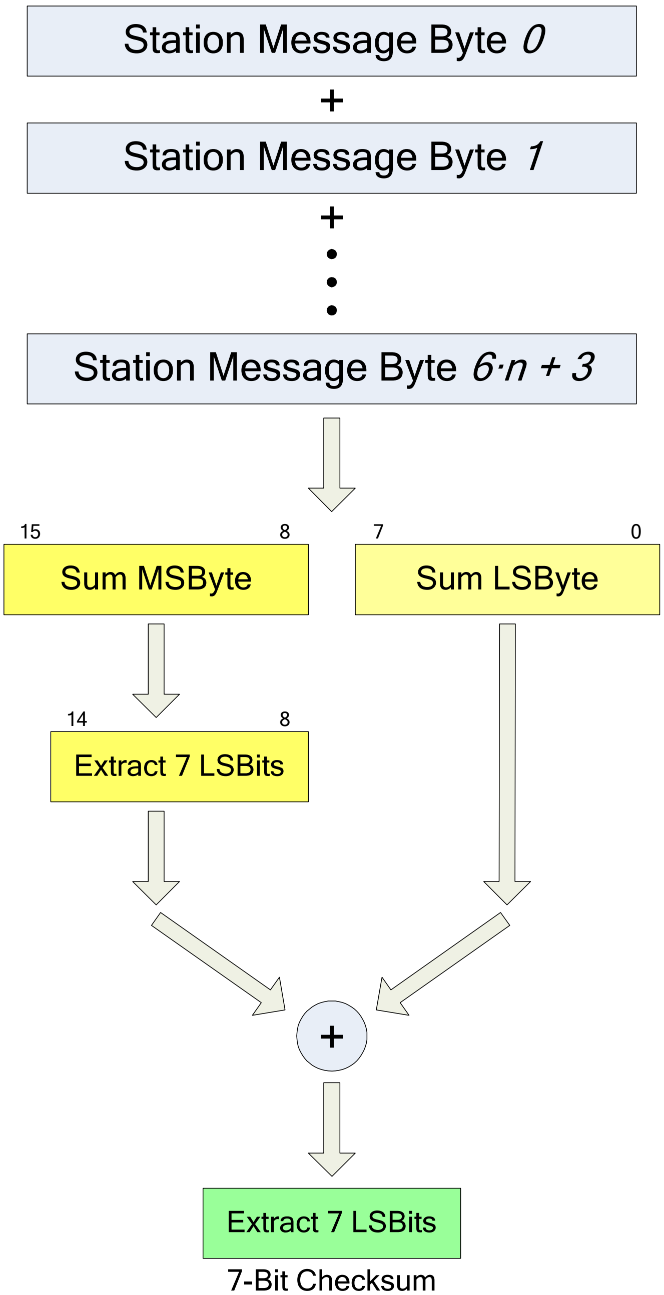


Figure 4-7: Checksum Calculation

4.6 Service Information Message (MSG ID 0110)

This message type is used to indicate the available audio and data services independently. It allows for future expansion of the number of audio programs and types. It provides an indication of multi-channel audio features. It also indicates features for available data services. This allows the receiver to enable or disable the desired services.

Figure 4-8 shows the message structure for the Service Information Message. The message consists of 27 bits which include a Service Category identifier and Service Descriptors. These bits aid the receiver in faster searching/scanning for available and/or desired programs.

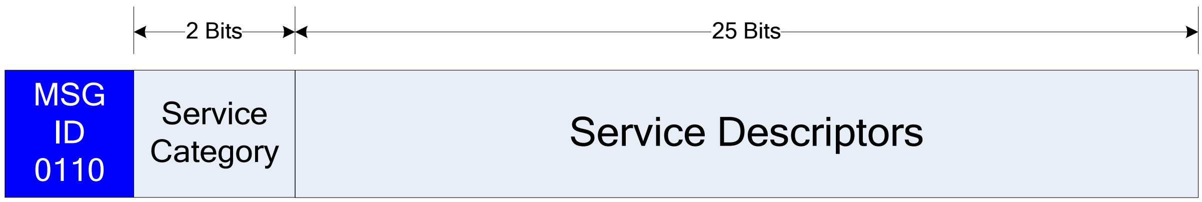


Figure 4-8: Service Information Message – Message Structure

The values for the Service Category identifier are shown in Table 4-9.

Table 4-9: Service Category Identifier – Values

Service Category Identifier Value	Service Category Identifier Description
00	Audio
01	Data
10 - 11	Reserved

The Service Descriptors for the audio and the data service categories are described in the following subsections.

4.6.1 Audio Service Descriptors

Figure 4-9 shows the Service Information Message structure containing descriptors for an audio program.

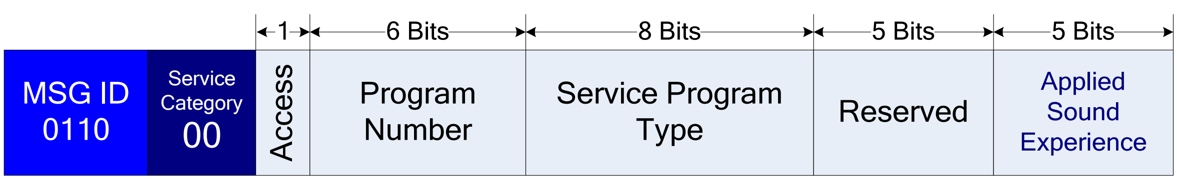


Figure 4-9: Service Information Message – Message Structure for Audio Service Descriptors

The audio service category is indicated by a service category identifier value of “00” in the Service Information Message.

The service descriptor portion contains information about Access, Program Number, Service Program Type, and Applied Sound Experience.

For the 1-bit Access descriptor, 0 indicates “public/unrestricted” and 1 indicates “restricted”.

The Program Number descriptor is used to identify and manage the MPS or SPS programs transmitted as shown in Table 4-10.

Table 4-10: Program Number Descriptor

Program Number	Description
0	MPS
1 - 7	SPS
8 - 63	Reserved

The Audio Service Program Types enable a receiver to search and sort through the variety of program content being broadcast. The Audio Service Program Types defined for use in the HD Radio system are described in IEC-62106-9 and are shown in Table 4-11. Audio Service Program Type numbers zero through 31 comply with Part 9 of the RDS specification (see the RDS Standard, IEC-62106-9).

NRSC Supplemental Information provides the most up-to-date information regarding the Audio Service Program Types: http://hdradio.com/broadcasters/us-regulatory/nrsc-supplemental-information.

Table 4-11: Audio Service Program Types

Audio Service Program Type Number	Audio Service Program Type Description
0	See the RDS Standard, IEC-62106-9
1	See the RDS Standard, IEC-62106-9
2	See the RDS Standard, IEC-62106-9
3	See the RDS Standard, IEC-62106-9
4	See the RDS Standard, IEC-62106-9
5	See the RDS Standard, IEC-62106-9
6	See the RDS Standard, IEC-62106-9
7	See the RDS Standard, IEC-62106-9
8	See the RDS Standard, IEC-62106-9
9	See the RDS Standard, IEC-62106-9
10	See the RDS Standard, IEC-62106-9
11	See the RDS Standard, IEC-62106-9
12	See the RDS Standard, IEC-62106-9
13	See the RDS Standard, IEC-62106-9
14	See the RDS Standard, IEC-62106-9
15	See the RDS Standard, IEC-62106-9
16	See the RDS Standard, IEC-62106-9
17	See the RDS Standard, IEC-62106-9
18	See the RDS Standard, IEC-62106-9
19	See the RDS Standard, IEC-62106-9
20	See the RDS Standard, IEC-62106-9
21	See the RDS Standard, IEC-62106-9
22	See the RDS Standard, IEC-62106-9
23	See the RDS Standard, IEC-62106-9
24	See the RDS Standard, IEC-62106-9
25	See the RDS Standard, IEC-62106-9
26	See the RDS Standard, IEC-62106-9
27	See the RDS Standard, IEC-62106-9
28	See the RDS Standard, IEC-62106-9
29	See the RDS Standard, IEC-62106-9
30	See the RDS Standard, IEC-62106-9

Audio Service Program Type Number	Audio Service Program Type Description
31	See the RDS Standard, IEC-62106-9
32 to 64	Reserved
65	Traffic
66 to 75	Reserved
76	Special Reading Services
77 to 255	Reserved

The bits in the Reserved descriptor default to “0”.

The Applied Sound Experience descriptor is used to indicate the type of sound and audio processing used. The various sound and audio processing methods used and their corresponding values in the Applied Sound Experience descriptor are shown in Table 4-12.

NRSC Supplemental Information provides the most up-to-date information regarding the Applied Sound Experience Types: http://hdradio.com/broadcasters/us-regulatory/nrsc-supplemental-information.

Table 4-12: Applied Sound Experience Descriptor

Applied Sound Experience Value	Applied Sound Experience Method
0	None
1	Reserved
2	Dolby Pro Logic II Surround
3	DTS Neural Surround
4	FhG MP3 Surround
5	DTS Neo:6 Surround
6	Reserved
7	DTS Neural:X Surround
8	Dolby Pro Logic IIx Surround
9	Dolby Pro Logic IIz Surround
10 to 31	Reserved

4.6.2 Data Service Descriptors

Figure 4-10 shows the Service Information Message structure containing descriptors for a data service.

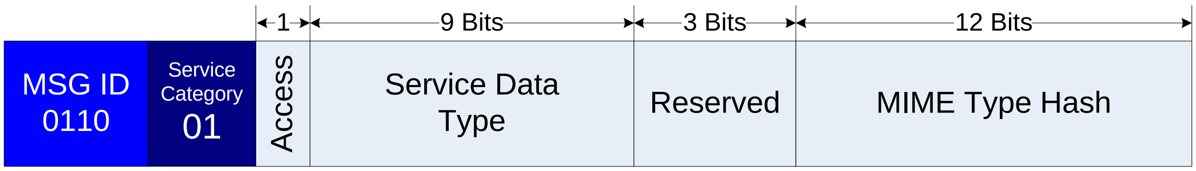


Figure 4-10: Service Information Message – Message Structure for Data Service Descriptors

The data service category is indicated by a service category identifier value of "01" in the Service Information Message.

The service descriptor portion contains information about Access, Service Data Type, and MIME Type Hash (only 12 least significant bits).

For the 1-bit Access descriptor, 0 indicates "public/unrestricted" and 1 indicates "restricted".

The Service Data Type descriptor is defined in Table 4-13.

NRSC Supplemental Information provides the most up-to-date information regarding the Service Data Types: http://hdradio.com/broadcasters/us-regulatory/nrsc-supplemental-information.

Table 4-13: Service Data Type Descriptor

Service Data Type Number	Service Data Type Definition
0	Non-specific
1	News
2	Reserved
3	Sports
4 to 28	Reserved
29	Weather
30	Reserved
31	Emergency
32 to 64	Reserved
65	Traffic
66	Image Maps
67 to 79	Reserved
80	Text
81 to 255	Reserved
256	Advertising
257	Financial
258	Stock Ticker
259	Navigation
260	Electronic Program Guide (EPG)
261	Audio
262	Private Data Network
263	Service Maintenance
264	HD Radio System Services

Service Data Type Number	Service Data Type Definition
265	Audio-Related Objects
266 to 510	Reserved
511	Reserved for Special Tests

The MIME Type Hash descriptor is used to indicate the type of Application MIME Type used.

4.7 SIS Parameter Message (MSG ID 0111)

This message type is used to carry various system parameters. The SIS Parameter Message has a payload of 22 bits which consists of a 6-bit index field and a 16-bit parameter field.

Figure 4-11 shows the message structure for the SIS Parameter Message.

Figure 4-11: SIS Parameter Message – Message Structure

The SIS Parameter Message indices are defined in Table 4-14.

Table 4-14: SIS Parameter Message Indices

Index	Description	Comments	Reference Figure
0	Leap Second Offset Most significant byte: Pending Offset (8-bit signed) Least significant byte: Current Offset (8-bit signed) Refer to Subsection 5.4 for details on the application of the Leap Second	Reference [21] If this information is unavailable, it is recommended that a default value of 0x12 12 be used, indicating that 18 leap seconds are current as well as pending. See Subsection 5.4.1 for details.	Figure 4-12
1	ALFN representing the GPS time of a pending leap second adjustment (16 LSBs)	Set to 0 if a leap second is not pending	Figure 4-13
2	ALFN representing the GPS time of a pending leap second adjustment (16 MSBs)	Set to 0 if a leap second is not pending	Figure 4-14
3	Local Time Data (Refer to Subsection 4.7.1) Refer to Subsection 5.4 for details on the application of the Local Time parameters.		Figure 4-15
4	Exciter Manufacturer ID / ICB (Importer Connected Bit)	Two 7-bit characters ISO/IEC 8859-1 Format Valid Values: 32 to 126	Figure 4-16
5	Exciter Core Version Number Levels 1 through 3	Level 1: Left-most (most significant level) Level 3: Right-most (least significant level) Valid Range: 0 to 30 31: Invalid / not used	Figure 4-17

Index	Description	Comments	Reference Figure
6	Exciter Manufacturer-assigned Version Number Levels 1 through 3	Level 1: Left-most (most significant level) Level 3: Right-most (least significant level) Valid Range: 0 to 30 31: Invalid / not used	Figure 4-18
7	Exciter Core Version Number 4 and Status / Exciter Manufacturer-assigned Version Number 4 and Status	Level 4 Version Number Valid Range: 0 to 30 31: Invalid / Not Used Status 0: Commercial Release 1: Engineering Release 2: Patch 3 to 7: Reserved	Figure 4-19
8	Importer Manufacturer ID	Two characters ISO/IEC 8859-1 Format Valid Values: 32 to 126	Figure 4-20
9	Importer Core Version Number Levels 1 through 3	Level 1: Left-most (most significant level) Level 3: Right-most (least significant level) Valid Range: 0 to 30 31: Invalid / not used	Figure 4-21
10	Importer Manufacturer-assigned Version Number Levels 1 through 3	Level 1: Left-most (most significant level) Level 3: Right-most (least significant level) Valid Range: 0 to 30 31: Invalid / not used	Figure 4-22
11	Importer Core Version Number 4 and Status / Importer Manufacturer-assigned Version Number 4 and Status	Level 4 Version Number Valid Range: 0 to 30 31: Invalid / Not Used Status 0: Commercial Release 1: Engineering Release 2: Patch 3 to 7: Reserved	Figure 4-23
12	Importer Configuration Number	Defines the Importer Audio and Data Service Configuration Range: 0 to 65535	Table 4-24

Index	Description	Comments	Reference Figure
13 to 63	Reserved for future use.	—	—

For each of the individual fields shown in Figure 4-12 through Figure 4-24, the most significant bit is on the left.

Figure 4-12: Format of SIS Parameter Message Index 0

Figure 4-13: Format of SIS Parameter Message Index 1

Figure 4-14: Format of SIS Parameter Message Index 2

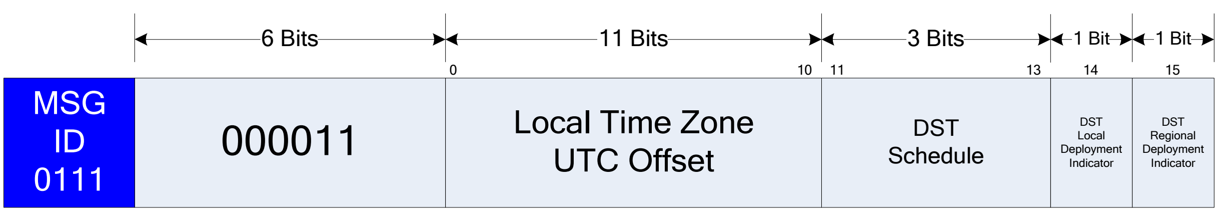


Figure 4-15: Format of SIS Parameter Message Index 3

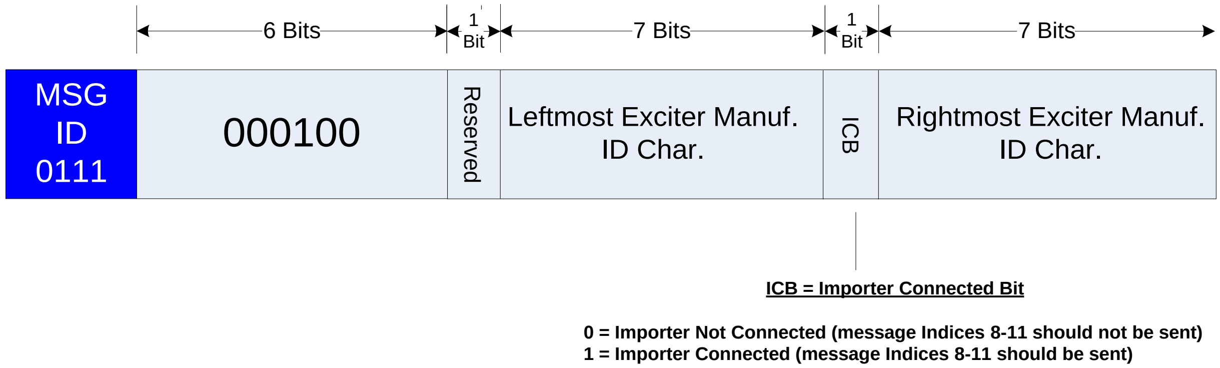


Figure 4-16: Format of SIS Parameter Message Index 4

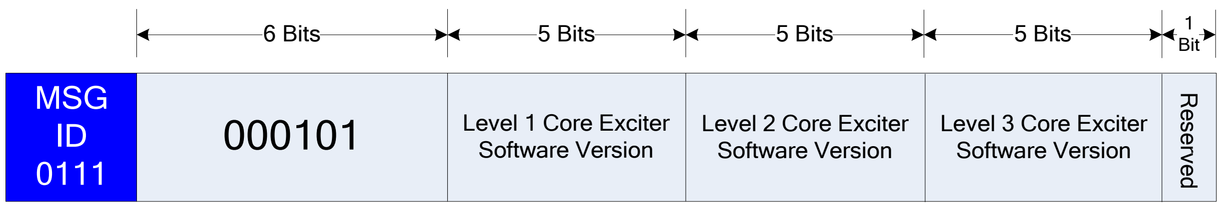


Figure 4-17: Format of SIS Parameter Message Index 5

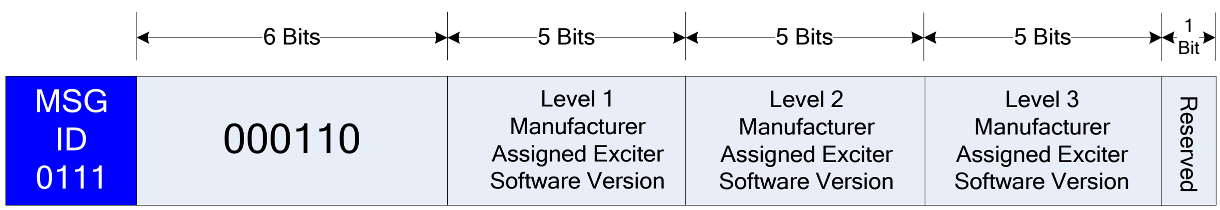


Figure 4-18: Format of SIS Parameter Message Index 6

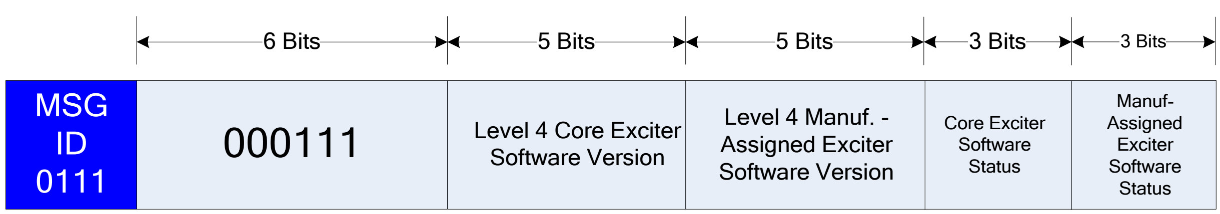


Figure 4-19: Format of SIS Parameter Message Index 7

Figure 4-20: Format of SIS Parameter Message Index 8

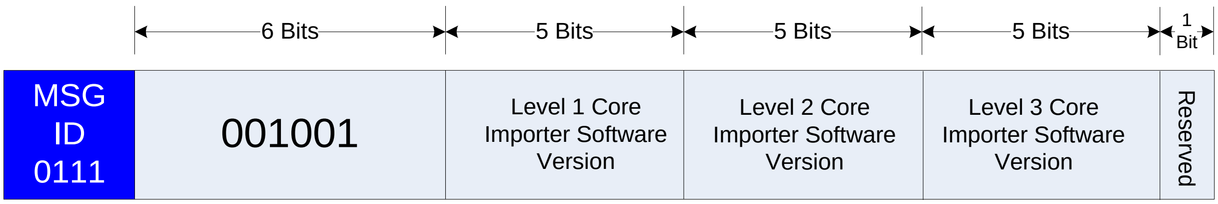


Figure 4-21: Format of SIS Parameter Message Index 9

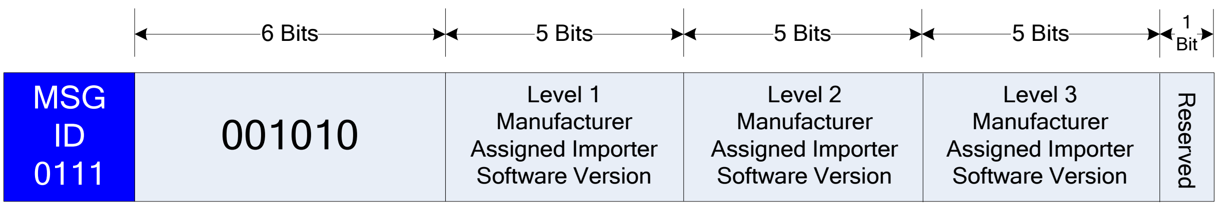


Figure 4-22: Format of SIS Parameter Message Index 10

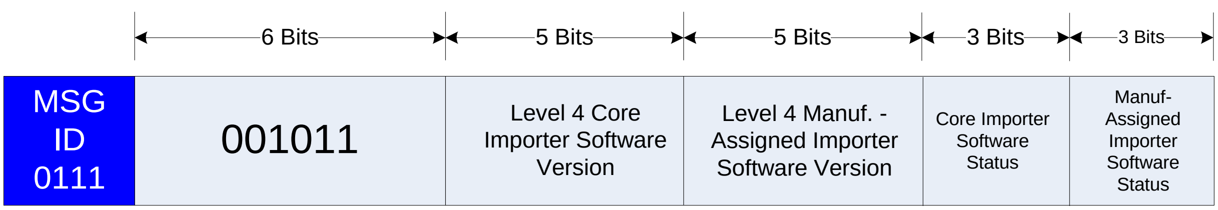


Figure 4-23: Format of SIS Parameter Message Index 11

	6 Bits	16 Bits
MSG ID 0111	001100	Importer Configuration Number

Figure 4-24: Format of SIS Parameter Message Index 12

The following subsections define the SIS Parameter Messages.

4.7.1 Local Time Parameters

The bit positions for the various Local Time parameters are summarized in Table 4-15.

Table 4-15: Local Time Parameters – Bit Positions

Bits	Parameter	Units	Format
0:10	Local Time Zone UTC Offset (See Subsection 4.7.1.1)	Minutes	Signed Integer
11:13	DST Schedule (See Subsection 4.7.1.2)	N/A	N/A
14	DST Local Deployment Indicator (See Subsection 4.7.1.3)	N/A	1 = yes
15	DST Regional Deployment Indicator (See Subsection 4.7.1.4)	N/A	1 = yes

4.7.1.1 Parameter – Local Time Zone UTC Offset

These bits constitute the higher order part of the word and provides static data on the local time zone (offset from UTC when DST not in effect, in minutes).

Data for the U.S. standard time zones are shown in Table 4-16. Time laws in the U.S. are the responsibility of the Department of Transportation (Reference [15]).

Table 4-16: Local Time Zone UTC Offset – U.S. Standard Time Zones

Time Zone Name	Bits (0:10)	UTC Reference
Atlantic	11100010000	(UTC –4 hours)
Eastern	11011010100	(UTC –5 hours)
Central	11010011000	(UTC –6 hours)
Mountain	11001011100	(UTC –7 hours)
Pacific	11000100000	(UTC –8 hours)
Alaska	10111100100	(UTC –9 hours)
Hawaii-Aleutian	10110101000	(UTC –10 hours)
Samoa	10101101100	(UTC –11 hours)
Chamorro	01001011000	(UTC +10 hours)
In addition, for Canada:		
Newfoundland	11100101110	(UTC –3 1/2 hours)

4.7.1.2 Parameter – DST Schedule

These bits provide static data on the schedule of Daylight Saving Time (DST) used regionally (for example, nationally), regardless of whether or not DST is practiced locally.

Table 4-17 shows the values of the bits 11:13 based on the Daylight Saving Time Schedule.

The dates and times in Table 4-17 are shown for reference; refer to the proper governing organization for the latest information (References [19] and [20]).

Table 4-17: DST Schedule

Bits 11:13	Daylight Saving Time Schedule
000	Daylight Saving Time not practiced in this nation (e.g., Japan, Central America) or Daylight Saving Time practiced on an irregular schedule (e.g., Israel, Palestine)
001	U.S./Canada Begins 2:00 AM on the second Sunday of March Ends 2:00 AM on the first Sunday of November Subject to change according to Reference [19]
010	European Union (EU): 01:00 UTC on last Sunday of March until 01:00 UTC on last Sunday of October. Subject to change according to Reference [20]
011 - 111	Reserved

Other global practices for DST may be added to this table as HD Radio broadcasts and receivers are introduced into those nations. An overflow of this table may be continued in another, future, SIS Parameter Message type.

In the United States and Canada, broadcasters must use a field value of 001 year-round, regardless of whether or not their local community practices Daylight Saving Time.

4.7.1.3 Parameter – DST Local Deployment Indicator

This bit provides static data on whether or not DST is practiced locally; 1 if it is and 0 if it is not.

In the United States, bit 14 is set to 1 year-round, except in Hawaii, American Samoa, Guam, Puerto Rico, the Virgin Islands, and major portions of Indiana and Arizona. In Canada, bit 14 is set to 1 year-round, except in most of Saskatchewan and portions of other Provinces, including British Columbia and Quebec.

The Energy Policy Act of 2005 extended Daylight Saving Time in the U.S. beginning in 2007, though Congress retained the right to revert to the 1986 law should the change prove unpopular or if energy savings are not significant. Going from 2007 forward, Daylight Saving Time in the U.S. begins at 2:00 a.m. on the second Sunday of March and ends at 2:00 a.m. on the first Sunday of November. See Reference [16].

Concurrence would require the use of UTC local offset 11001011100 (UTC – 7 hours), DST Schedule 001, and DST Practice (Bit 14) 0, year-round, in order to make California broadcasters most compatible with those of neighboring communities (e.g., Yuma, AZ).

Information regarding DST in this AIDD is subject to change, according to Reference [19].

4.7.1.4 Parameter – DST Regional Deployment Indicator

This bit provides seasonal data as to whether or not DST is in effect regionally (for example, nationally); 1 if it is and 0 if it is not.

Simple receivers can use this bit exclusively (ignoring Bits 11:13) to determine when to set the display clock one hour forward. Receivers should honor this bit only if Bit 14 or user setup indicates that DST is practiced locally. However, since this datum is not guaranteed in real time (either by all broadcasters, or in a timely manner by broadcasters that do provide it), more upscale receiver designs may instead prefer to internally compute the period of DST using the static data provided in bits 11:13. This will provide a better consumer experience. However, all receivers should honor this bit in preference to any predetermined schedule indicated by Bits 11:13. National rules for DST change occasionally and the receiver firmware may not be up-to-date or appropriate for the nation in which the receiver is being used.

In the United States and Canada, this bit should be set to 1 when the nation as a whole is practicing Daylight Saving Time (that is, in the summer), regardless of whether or not it is being practiced locally.

4.7.2 Broadcast Equipment Software Version Information

Message indices 4 through 12 may be utilized to broadcast software version information. Such information may be useful for tracking the evolution of the system and understanding field operation conditions. Such information may also be useful in aiding receivers to identify stations that support new features as the system evolves.

Message index 5 identifies the Exciter Core Software Version Number. The Core Software Version Number is independent of the specific equipment manufacturer.

Message index 4 provides a two-character Manufacturer ID identifying the specific equipment manufacturer.

Message index 6 provides the manufacturer-assigned version number for the Exciter.

If the broadcast system includes an Importer, the Importer Connected Bit (ICB) shall be set to “1” and message indices 8 through 12 shall be broadcast to convey the Importer software information. If the broadcast system does not include an Importer, the ICB shall be “0” and message indices 8 through 12 shall be omitted.

4.8 Universal Short Station Name / Station Slogan (MSG ID 1000)

This message type conveys either the Universal Short Station Name or the Station Slogan, depending on the state of the Name Type bit embedded in the message structure. Subsection 4.8.1 provides the details for the Universal Short Station Name; Subsection 4.8.2 provides the details for the Station Slogan.

Refer to Figure 4-25 for an illustration of the general message structure for this message type and Table 4-18 for a description of the fields in this common message structure.

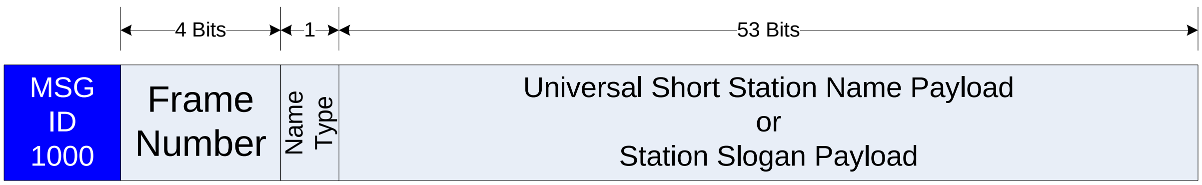


Figure 4-25: General Message Structure for MSG ID 1000

Table 4-18: Description of Fields within the Common Message Structure for MSG ID 1000

Field Name	Range	Description
Name Type	0 - 1	Type = 0 indicates that the message type is Universal Short Station Name Type = 1 indicates that the message type is Station Slogan
Frame Number	If Name Type = 0, Range = 0 - 1 If Name Type = 1, Range = 0 - 15	Indicates the current frame number of the message. Always zero for single-frame messages.

4.8.1 Universal Short Station Name

The Universal Short Station Name provides additional capabilities beyond the standard Short Station Name (MSG ID 0001) message type. The Universal Short Station Name may be up to 12 characters in length, as shown in Figure 4-26. In addition, it supports international character sets.

The Universal Short Station Name cannot be broadcast in the SIS concurrently with a standard Short Station Name. If the standard Short Station Name is sufficient, it is recommended that it be used exclusively, due to its higher efficiency. However, for applications requiring more than four characters in the station name, such as weather/FEMA stations (760 associated stations) or Low Power FM (LPFM) stations which use five or six alphanumerical characters, the Universal Short Station Name must be used. Also, for countries outside the U.S., the Universal Short Station Name may be required if the standard Short Station Name character set is insufficient and/or the station name is longer than four characters.

The Universal Short Station Name message structure is shown in Figure 4-26. The format allows for up to two frames of character data to be sent, which provides for 12 8-bit characters or six 16-bit characters. The format of the second frame is different from the first, as shown. Table 4-19 and Table 4-20 describe the data fields for the first and second frames, respectively.

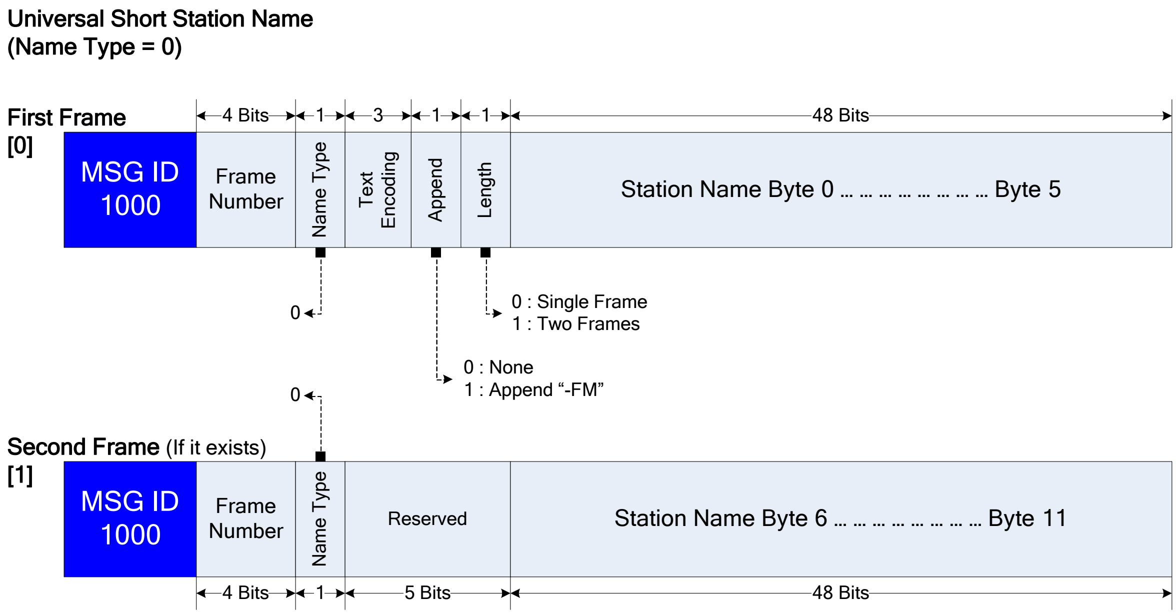


Figure 4-26: Universal Short Station Name – Message Structure

Table 4-19: Description of Universal Short Station Name Fields for Frame Number = 0

Field Name	Range	Description
Text Encoding	0 to 7	See Table 4-8
Append	0 to 1	Append = 0: Receiver shall not append any information after the Station Name display Append = 1: Receiver shall append "-FM" after the Station Name display
Length	0 to 1	Length = 0: Station Name contained in a single frame Length = 1: Station Name contained in two frames
Station Name	N/A	For 8-bit character encoding, frame 0 contains six characters of the Station Name Byte 0 is the leftmost character For single-frame Station Names, any unused bytes to the right of the Station Name text are filled with NULL characters (0x00) For 16-bit character encoding, frame 0 contains three characters of the Station Name Bytes 0:1 convey the leftmost character For single-frame Station Names, any unused byte pairs to the right of the Station Name text are filled with NULL characters (0x00 00)

Table 4-20: Description of Universal Short Station Name Fields for Frame Number = 1

Field Name	Range	Description
Station Name	N/A	For 8-bit character encoding, frame 1 contains the remaining characters of the Station Name Byte 11 is the rightmost character Any unused bytes to the right of the Station Name text are filled with NULL characters (0x00) For 16-bit character encoding, frame 1 contains the remaining characters of the Station Name Bytes 10:11 convey the rightmost character Any unused byte pairs to the right of the Station Name text are filled with NULL characters (0x00 00)

4.8.2 Station Slogan

The Station Slogan message structure is shown in Figure 4-27. The format allows for up to 16 frames of character data to be sent, which accommodates up to 95 8-bit characters or 47 16-bit characters. The format of the first Station Slogan frame is different from the others, as shown. Table 4-21 and Table 4-22 describe the data fields for the first and subsequent frames, respectively.

When the Station Slogan is broadcast in conjunction with the Long Station Name (MSG ID 0010), both messages must have the same content.

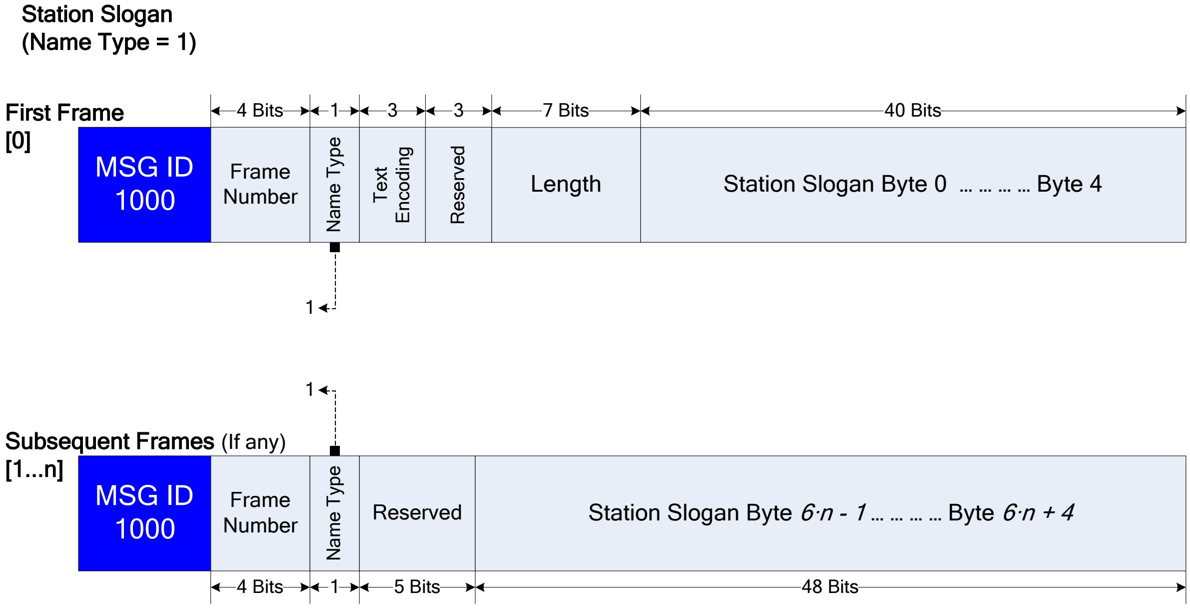


Figure 4-27: Station Slogan – Message Structure

Table 4-21: Description of Station Slogan Fields for Frame Number = 0

Field Name	Range	Description
Text Encoding	0 to 7	See Table 4-8
Length	5 to 95, 8-bit encoding 4 to 94, 16-bit encoding	Defines the total number of bytes of the Station Slogan text, excluding any unused bytes in the last frame For 16-bit character encoding, the Length must be even.
Station Slogan	N/A	For 8-bit character encoding, frame 0 contains the first 5 characters of the Station Slogan Byte 0 is the leftmost character For single-frame Station Slogans, any unused bytes to the right of the Station Slogan text are filled with NULL characters (0x00) For 16-bit character encoding, frame 0 contains the first two characters of the Station Slogan Bytes 0:1 convey the leftmost character Byte 4 is always set to zero (0x00) For single-frame Station Slogans, any unused byte pairs to the right of the Station Slogan text are filled with NULL characters (0x00 00)

Table 4-22: Description of Station Slogan Fields for Frame Number = 1 to n

Field Name	Range	Description
Station Slogan	N/A	For 8-bit character encoding, frames 1 to n contain the additional characters of the Station Slogan, where the lowest numbered byte within a frame is the leftmost for that frame For the last frame, any unused bytes to the right of the Station Slogan text are filled with NULL characters (0x00) For 16-bit character encoding, frames 1 to n contain the additional characters of the Station Slogan, where the lowest numbered byte-pair within a frame is the leftmost for that frame For the last frame, any unused byte pairs to the right of the Station Slogan text are filled with NULL characters (0x00 00)

4.9 Emergency Alerts Message (MSG ID 1001)

This message type allows the station to send alert messages. The alert messages are primarily intended for emergency public alerting, and the alert messages address any cause that may be defined by CAP [31]. Examples include weather alerts, child abduction alerts (AMBER alerts), and HAZMAT alerts. The message allows for waking-up receivers and further intelligent handling of the information provided by the station.

Emergency Alerts are conveyed in the message type and are denoted in this document as EA Messages. The EA Message has a total payload of 58 bits per frame. It can span from two (minimum) frames to 64 frames. Figure 4-28 shows the message structure for the Emergency Alerts Message. The format of the first frame is different from the others, as shown.

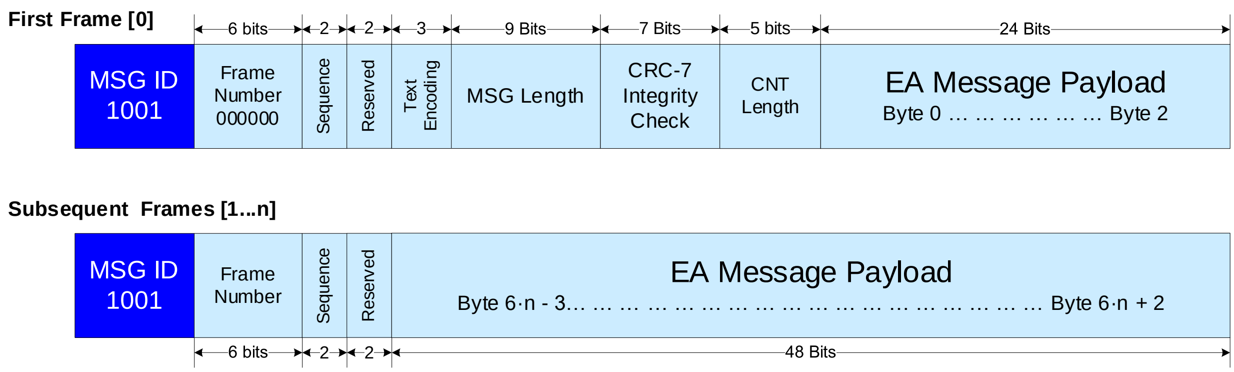


Figure 4-28: Emergency Alerts Message – Message Structure

The Emergency Alerts Message can be used to send a string of up to 381 bytes per message which may include various bit-oriented data elements as well as 8-bit characters [22] or 16-bit characters [23]. A message may span up to 64 frames. Each message contains a sequence number indicating when the message content has changed. Any change in the message content is considered a new message and the sequence number is incremented. Length information and a 7-bit integrity check are included in the first frame to increase receive reliability.

Table 4-23 and Table 4-24 describe the data fields for the first and subsequent frames, respectively.

Table 4-23: Description of EA Message Fields for Frame Number = 0

Field Name	Range	Description
Frame Number	0 to 63	Indicates the current frame number of the message Set to zero for the first frame
Sequence	0 to 3	Increments by 1, modulo 4, whenever the EA Message content changes A new sequence number must commence with Frame 0 and the same number shall be used for all frames of a given EA Message
Reserved	0 to 3	Reserved for future use
Text Encoding	0 to 7	See Table 4-25
MSG Length	7 to 381	Defines the total number of bytes of the EA Message Payload, excluding any unused bytes in the last frame
CRC-7 Integrity Check	0 to 127	CRC-7 computation of all the bytes of the EA Message Payload, as indicated by ‘MSG Length’
CNT Length	3 to 31	Defines the total number of byte-pairs of the control (CNT) data included in the EA Message Payload

Field Name	Range	Description
EA Message Payload	N/A	Frame 0 contains the first three bytes of the EA Message Payload. Byte 0 is the leftmost byte These three bytes contain control data.

Table 4-24: Description of Station Message Fields for Frame Number = 1 to n

Field Name	Range	Description
Frame Number	1 to 63	Indicates the current frame number of the message
Sequence	0 to 3	Increments by 1, modulo 4, whenever the EA Message content changes A new sequence number must commence with Frame 0 and the same number shall be used for all frames of a given EA Message
Reserved	0 to 7	Reserved for future use
EA Message Payload	N/A	Frames 1 to 63 contain bytes 4 to 380 of the EA Message Payload. The count is from the leftmost byte. These bytes contain control data and may contain text when conveyed in the message. In the last frame of an EA Message of any length, any unused bytes to the right of the EA Message Payload are filled with NULL characters (0x00)

NRSC Supplemental Information provides the most up-to-date information regarding the Text Encoding Definitions: http://hdradio.com/broadcasters/us-regulatory/nrsc-supplemental-information.

Table 4-25: Text Encoding Definitions – Emergency Alerts

Value	Field Definition
000 (default)	ISO/IEC 8859-1:1998 (Reference [22])
001	Reserved
010 and 011	Reserved
100	ISO/IEC 10646-1:2000 UCS-2 (Little Endian) (Reference [23])
101 to 111	Reserved

4.9.1 EA Message Schedule Requirements

Regularly, scheduling SIS PDU messages may be configured by a broadcaster, based on various considerations: otherwise, it may use the default scheduling as further indicated in this document.

However, when broadcasting EA Messages for the purpose of emergency public alerting, it is necessary to guarantee that broadcasting a message is completed within a specific time. This is expected by receivers, which are monitoring for alert messages, while also attempting to minimize their power consumption, whether in standby (ready to wake-up) or turned ‘On’.

In order to avoid inadvertent scheduling when broadcasting alert messages and to provide adequate user experience, fixed (non-configurable) scheduling for SIS PDU messages is employed when broadcasting an EA Message. For details refer to Section 5.2.

4.10 Advanced Service Information Message (MSG ID 1010)

This message type is used to indicate the available audio and data services supported by the P4 channel in Layer 1 advanced service modes MP1X and MP1XOV. Its definition is identical to that of MSG ID 0110 – Service Information Message, except that it utilizes a different MSG ID to ensure backward compatibility of MP1X and MP1XOV with legacy receivers. Only newer, non-legacy receivers are able to decode Advanced Service Information Messages. Legacy receivers will ignore this SIS message type.

Figure 4-29 shows the message structure for the Advanced Service Information Message. The message consists of 27 bits which include a Service Category identifier and Service Descriptors. These bits aid the receiver in faster searching/scanning for available and/or desired programs. Refer to Subsection 4.6 for details.

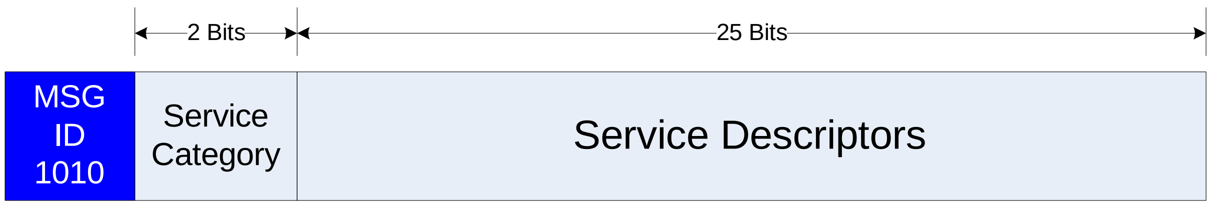


Figure 4-29: Advanced Service Information Message – Message Structure

4.11 CRC Field

Each PDU is terminated with a 12-bit Cyclic Redundancy Check (CRC) for the purpose of aiding the receiver in detecting transmission errors. The CRC, ordered as PDU bits 79:68, is computed as follows:

1. Fill PDU bits 79:68 with zeros.
2. Perform modulo-two division of PDU bits 79:0 by the generator polynomial g(x),

   Where $g(x) = X^{12} + X^{11} + X^{3} + X + 1$

   and PDU bit 79 is computed first.
3. The 12-bit remainder is then copied back into PDU bits 68:79, where bit 68 is considered the most significant remainder bit and bit 79 is the least significant remainder bit.

5 Applications and Examples

5.1 Station Location Example

As an example of how the position information is constructed, consider a location at N 39˚ 11’ 46.32”, W 76˚ 49’ 6.59’, and an altitude of 90.7 meters. The first step is to convert latitude and longitude to decimal degrees:

Latitude = $39 + \frac{11}{60} + \frac{46.32}{3600} = 39.1962 \text{ deg}$

Longitude = $76 + \frac{49}{60} + \frac{6.59}{3600} = 76.8185 \text{ deg}$

The next step is to convert all three parameters to the proper fractional format:

Latitude: 39.1962˚ · 8192 = 321095 rounded to the nearest integer

= 0b0001001110011001000111 (22-bit binary)

Longitude: 76.8185˚ · 8192 = 629297 rounded to the nearest integer

= 0b0010011001101000110001 (22-bit binary)

However, it is necessary to take the two’s complement of this number to get west longitude

West longitude = 0b1101100110010111001111 (22-bit binary)

Altitude = ROUND (90.7/16) = 6 = 0b0110 = 0x06

Finally, the parameters are then packed into the appropriate message format:

Station Location (High Portion):

Append a “1” in the most significant bit position of the latitude:

= 0b10001001110011001000111 (23-bit binary)

Append the upper four bits of altitude in the four least significant bit positions:

= 0b100010011100110010001110000 (27-bit binary)

Or in hex notation = 0x89CC8E0 (left-justified)

Station Location (Low Portion):

Append a “0” in the most significant bit position of the longitude:

= 0b01101100110010111001111 (23-bit binary)

Append the lower four bits of altitude in the four least significant bit positions:

= 0b011011001100101110011110110 (27-bit binary)

Or in hex notation = 0x6CCB9EC (left-justified)

It must be noted that frequency translator stations must exercise care in the use of the station ID number, station name, and station location. If the translator station acts as a repeater, then it will convey the station information of the primary station, not the translator station. It may be necessary for the translator station to produce its own station information to ensure proper operation of the system.

5.2 Scheduling of SIS PDUs on the PIDS Logical Channel

PDU scheduling on a block basis is provided for various cases. The schedule repeats every L1 frame. The FM system broadcasts 16 PIDS blocks per L1 frame while the AM system broadcasts only eight PIDS blocks per L1 frame. The following SIS message types are required to be sent in order to guarantee compatibility with all receivers:

- ID = 0 Station ID: Used for receiver service following and is scheduled often in order to optimize receiver scanning time. For details on how this schedule may be utilized for TPEG Traffic Services, refer to [33].
- ID = 4 Station Location
- ID = 6/10 Service Info / Advanced Service Info Message: Used for receiver data service acquisition and is scheduled often in order to optimize receiver scanning time. For details on how this schedule may be utilized for TPEG Traffic Services, refer to [33].
- ID = 7 SIS Parameter Message
- ID = 1 Short Station Name OR ID = 8 Universal Short Station Name: that is, a valid call sign must be sent. Note that it must be one or the other. Both message types cannot be sent as part of the same schedule.

In addition, if the EA system is active, then Message ID 1001 (Emergency Alerts) is also mandatory. When the station broadcasts an EA Message for conveying an Emergency Public Alert, it takes the highest priority, and the SIS schedule is optimized accordingly. The schedule allows for the broadcast of a complete EA Message in less than eight seconds in FM and in less than 16 seconds in AM.

Based on all of this information, there are a total of eight cases to consider for SIS scheduling, depending on the state of each of the following:

- Band: AM / FM
- EA Active / EA Inactive
- Short Station Name / Universal Short Station Name

The schedules are shown in Table 5-1 to Table 5-4 for FM and Table 5-5 to Table 5-8 for AM. Note that for all of these scenarios, all possible non-required message types are provided for in the schedules. Such message types all convey textual information that may be useful to the consumer, such as station message or station slogan. However, if a broadcaster chooses not to utilize one of these textual message types, it shall be removed from the schedule and replaced with another message type. Empty text strings shall not be broadcast. Broadcast of station slogan is highly recommended as it provides the consumer with an understandable means of identifying each station.

In each scheduling scenario, one PIDS block is allocated for sending SIS Parameter Messages. Since this information is not time-critical, the broadcast system should cycle through each message, one at a time, whenever a SIS Parameter Message is scheduled. Since there are a total of 12 SIS Parameter Messages defined, a cycle will require 12 L1 frames (equivalent to approximately 18 seconds).

5.2.1 FM Scheduling

The FM SIS schedules are shown in Table 5-1 to Table 5-4. The schedules are optimized first for efficiency of the EA service (if enabled) and secondly for supporting data fast acquisition and service following. In the U.S., when EA is not active, the SIS schedule of Table 5-1 shall be used. When EA is active, all non-mandatory message types are removed from the schedule as shown in Table 5-2. As shown in Table 5-3, there is some loss in scheduling efficiency in order to provide for longer station names.

Many of the advanced service modes provide additional capacity in the PIDS logical channel and some of them also provide additional PIDSOV and/or SIDS logical channels. This capacity may be used to expand SIS beyond one PDU per block. It may also be used to support Low Latency Data Services (LLDS). Refer to [37] for details on LLDS and further information on how SIS and LLDS are multiplexed on the PIDS, PIDSOV, and SIDS logical channels. The scheduling information presented below should be considered the minimum SIS message schedule for advanced service modes.

In service Modes MP1X and MP1XOV, Service Info Messages describing the P4 logical channel services shall utilize MSG ID 10. However, for scheduling purposes MSG IDs 6 and 10 are considered to be the same message type and are scheduled as shown below.

Table 5-1: SIS Schedule – FM Band, Station name is ≤ 4 Characters, EA NOT Active

Block	Payload 1	Payload 2
0	SHORT STATION NAME (ID=1)	STATION ID (ID=0)
1	SERVICE INFO MESSAGE (ID=6/10)*	SERVICE INFO MESSAGE (ID=6/10)*
2	SHORT STATION NAME (ID=1)	STATION ID (ID=0)
3	STATION SLOGAN (ID=8)	
4	SHORT STATION NAME (ID=1)	STATION ID (ID=0)
5	SERVICE INFO MESSAGE (ID=6/10)*	SERVICE INFO MESSAGE (ID=6/10)*
6	SHORT STATION NAME (ID=1)	STATION ID (ID=0)
7	STATION LOCATION (ID=4)	STATION LOCATION (ID=4)
8	STATION MESSAGE (ID=5)	
9	SHORT STATION NAME (ID=1)	STATION ID (ID=0)
10	SERVICE INFO MESSAGE (ID=6/10)*	SERVICE INFO MESSAGE (ID=6/10)*
11	LONG STATION NAME (ID = 2)	
12	SIS PARAMETER MESSAGE (ID=7)	STATION ID (ID=0)
13	STATION MESSAGE (ID=5)	
14	SHORT STATION NAME (ID=1)	STATION ID (ID=0)
15	SERVICE INFO MESSAGE (ID=6/10)*	SERVICE INFO MESSAGE (ID=6/10)*

* NOTE
For transmission in Service Mode MP1X or MP1XOV, Service Info Messages describing the P4 logical channel services will utilize MSG ID 10.

Table 5-2: SIS Schedule – FM Band, Station name is ≤ 4 Characters, EA Active

Block	Payload 1	Payload 2
0	SHORT STATION NAME (ID=1)	STATION ID (ID=0)
1	EA MESSAGE (ID= 9)	
2	EA MESSAGE (ID= 9)	
3	STATION LOCATION (ID=4)	SERVICE INFO MESSAGE (ID=6/10)*
4	EA MESSAGE (ID= 9)	
5	EA MESSAGE (ID= 9)	
6	SIS PARAMETER MESSAGE (ID=7)	STATION ID (ID=0)
7	EA MESSAGE (ID= 9)	
8	SERVICE INFO MESSAGE (ID=6/10)*	SERVICE INFO MESSAGE (ID=6/10)*
9	EA MESSAGE (ID= 9)	
10	EA MESSAGE (ID= 9)	
11	SHORT STATION NAME (ID=1)	STATION ID (ID=0)
12	EA MESSAGE (ID= 9)	
13	EA MESSAGE (ID= 9)	
14	SHORT STATION NAME (ID=1)	SERVICE INFO MESSAGE (ID=6/10)*
15	EA MESSAGE (ID= 9)	

* NOTE
For transmission in Service Mode MP1X or MP1XOV, Service Info Messages describing the P4 logical channel services will utilize MSG ID 10.

Table 5-3: SIS Schedule – FM Band, Station name is 5 to 12 Characters, EA NOT Active

Block	Payload 1	Payload 2
0	UNIVERSAL SHORT STATION NAME (ID=8)	
1	SERVICE INFO MESSAGE (ID=6/10)*	SERVICE INFO MESSAGE (ID=6/10)*
2	STATION SLOGAN (ID=8)	
3	SIS PARAMETER MESSAGE (ID=7)	STATION ID (ID=0)
4	UNIVERSAL SHORT STATION NAME (ID=8)	
5	SERVICE INFO MESSAGE (ID=6/10)*	SERVICE INFO MESSAGE (ID=6/10)*
6	STATION LOCATION (ID=4)	STATION LOCATION (ID=4)
7	SIS PARAMETER MESSAGE (ID=7)	STATION ID (ID=0)
8	STATION MESSAGE (ID=5)	
9	SERVICE INFO MESSAGE (ID=6/10)*	SERVICE INFO MESSAGE (ID=6/10)*
10	UNIVERSAL SHORT STATION NAME (ID=8)	
11	STATION SLOGAN (ID=8)	
12	SIS PARAMETER MESSAGE (ID=7)	STATION ID (ID=0)
13	STATION MESSAGE (ID=5)	
14	SERVICE INFO MESSAGE (ID=6/10)*	SERVICE INFO MESSAGE (ID=6/10)*
15	SIS PARAMETER MESSAGE (ID=7)	STATION ID (ID=0)

* NOTE
For transmission in Service Mode MP1X or MP1XOV, Service Info Messages describing the P4 logical channel services will utilize MSG ID 10.

Table 5-4: SIS Schedule – FM Band, Station name is 5 to 12 Characters, EA Active

Block	Payload 1	Payload 2
0	UNIVERSAL SHORT STATION NAME (ID=8)	
1	EA MESSAGE (ID= 9)	
2	EA MESSAGE (ID= 9)	
3	STATION LOCATION (ID=4)	SERVICE INFO MESSAGE (ID=6/10)*
4	EA MESSAGE (ID= 9)	
5	EA MESSAGE (ID= 9)	
6	SIS PARAMETER MESSAGE (ID=7)	STATION ID (ID=0)
7	EA MESSAGE (ID= 9)	
8	SERVICE INFO MESSAGE (ID=6/10)*	SERVICE INFO MESSAGE (ID=6/10)*
9	EA MESSAGE (ID= 9)	
10	EA MESSAGE (ID= 9)	
11	SIS PARAMETER MESSAGE (ID=7)	STATION ID (ID=0)
12	EA MESSAGE (ID= 9)	
13	EA MESSAGE (ID= 9)	
14	SERVICE INFO MESSAGE (ID=6/10)*	SERVICE INFO MESSAGE (ID=6/10)*
15	EA MESSAGE (ID= 9)	

* NOTE
For transmission in Service Mode MP1X or MP1XOV, Service Info Messages describing the P4 logical channel services will utilize MSG ID 10.

5.2.2 AM Scheduling

For the AM schedules, similar rules apply except that the schedules repeat every eight blocks instead of 16. Refer to Table 5-5 to Table 5-8.

Table 5-5: SIS Schedule – AM Band, Station name is ≤ 4 Characters, EA NOT active

Block	Payload 1	Payload 2
0	SHORT STATION NAME (ID=1)	STATION ID (ID=0)
1	STATION MESSAGE (ID=5)	
2	SERVICE INFO MESSAGE (ID=6)	SHORT STATION NAME (ID=1)
3	SIS PARAMETER MESSAGE (ID=7)	STATION LOCATION (ID=4)
4	SHORT STATION NAME (ID=1)	STATION ID (ID=0)
5	STATION SLOGAN (ID=8)	
6	SERVICE INFO MESSAGE (ID=6)	SHORT STATION NAME (ID=1)
7	LONG STATION NAME (ID = 2)	

Table 5-6: SIS Schedule – AM Band, Station name is ≤ 4 Characters, EA Active

Block	Payload 1	Payload 2
0	SHORT STATION NAME (ID=1)	STATION ID (ID=0)
1	EA MESSAGE (ID= 9)	
2	EA MESSAGE (ID= 9)	
3	STATION LOCATION (ID=4)	SERVICE INFO MESSAGE (ID=6)
4	EA MESSAGE (ID= 9)	
5	EA MESSAGE (ID= 9)	
6	SIS PARAMETER MESSAGE (ID=7)	STATION ID (ID=0)
7	EA MESSAGE (ID= 9)	

Table 5-7: SIS Schedule – AM Band, Station name is 5 to 12 Characters, EA NOT Active

Block	Payload 1	Payload 2
0	UNIVERSAL SHORT STATION NAME (ID=8)	
1	STATION MESSAGE (ID=5)	
2	SERVICE INFO MESSAGE (ID=6)	STATION LOCATION (ID=4)
3	SIS PARAMETER MESSAGE (ID=7)	STATION ID (ID=0)
4	SERVICE INFO MESSAGE (ID=6)	SERVICE INFO MESSAGE (ID=6)
5	STATION SLOGAN (ID=8)	
6	SIS PARAMETER MESSAGE (ID=7)	STATION ID (ID=0)
7	SERVICE INFO MESSAGE (ID=6)	SERVICE INFO MESSAGE (ID=6)

Table 5-8: SIS Schedule – AM Band, Station name is 5 to 12 Characters, EA Active

Block	Payload 1	Payload 2
0	UNIVERSAL SHORT STATION NAME (ID=8)	
1	EA MESSAGE (ID= 9)	
2	EA MESSAGE (ID= 9)	
3	STATION LOCATION (ID=4)	SERVICE INFO MESSAGE (ID=6)
4	EA MESSAGE (ID= 9)	
5	EA MESSAGE (ID= 9)	
6	SIS PARAMETER MESSAGE (ID=7)	STATION ID (ID=0)
7	EA MESSAGE (ID= 9)	

5.3 Advanced Absolute L1 Frame Number Processing

The SIS Transport allocates two bits to broadcast the absolute L1 frame number in a serial fashion. The format is different for AM and FM as outlined in the following two subsections. In both cases, the value of ALFN to be transmitted over the PIDS channel is updated coincident with L1 block 0 of each L1 frame.

5.3.1 FM System Processing

The 16 LSBs, labeled d16 through d31in Figure 5-1, are transmitted as 2-bit pairs mapped into the ADV ALFN field of each PIDS block starting with block 0. ALFN bits d30:31 are broadcast at block 0 of each frame, ALFN bits d28:29 are broadcast at block 1 of each frame, and ALFN bits d16:17 are broadcast at block 7 of each frame.

ALFN bits d0:15 are further subdivided into pairs and mapped to the ADV ALFN field in blocks 8 through 15 as shown in Figure 5-1.

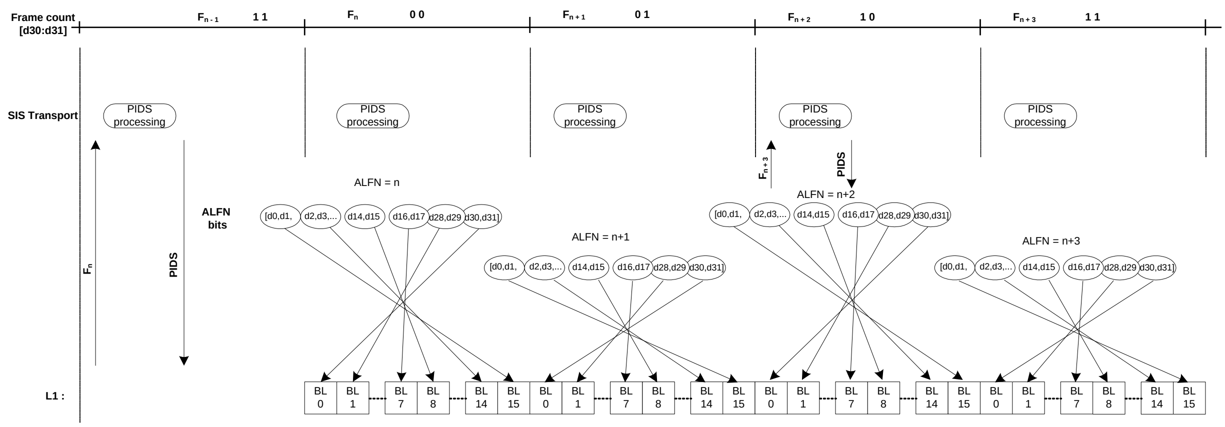


Figure 5-1: Broadcasting ALFN over the HD Radio FM System

5.3.2 AM System Processing

The 32 bits are subdivided into 16 bits numbered d16 through d31 (16 LSBs) and 16 bits indexed d0 through d15 (16 MSBs), as shown in Figure 5-2. ALFN bits d16:31 are subdivided into pairs and mapped to the two-bit ADV ALFN field of each PIDS block starting with block 0. ALFN bits d30:31 are broadcast at block 0 of each frame, ALFN bits d28:29 are broadcast at block 1 of each frame and ALFN bits d16:17 are broadcast at block 7 of the frame. This process takes place when ALFN d30:31 are not equal to 00.

ALFN bits d0:15 are subdivided into pairs and mapped to the ADV ALFN field in blocks 0 through 7 as shown. This occurs once with every four frames and is indicated when ALFN d30:31 are equal to 00.

The 16 LSBs of the ALFN are broadcast in three out of every four PIDS blocks; the 16 MSBs are broadcast once every four PIDS blocks.

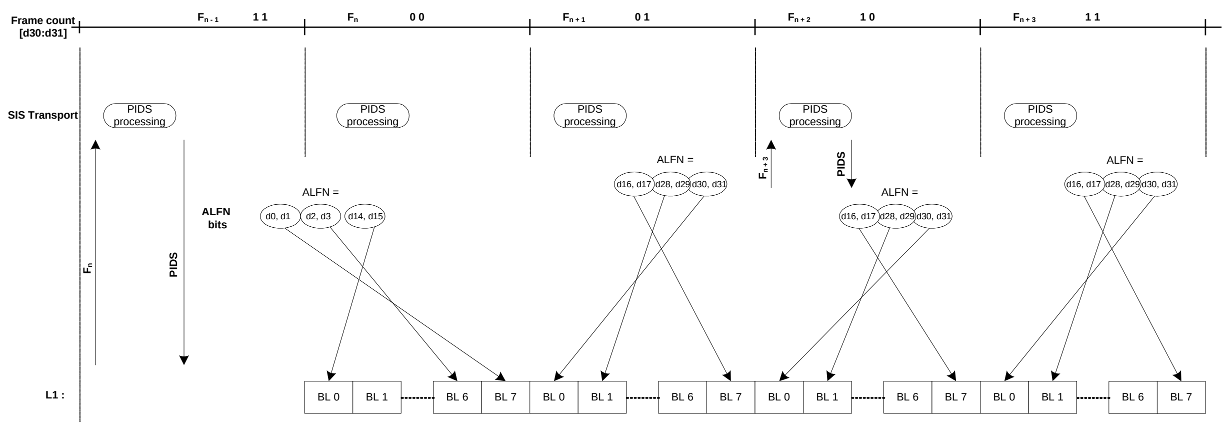


Figure 5-2: Broadcasting ALFN over the HD Radio AM System

5.3.3 Handling of Absolute L1 Frame Number in Layer 1

L1 does not handle ALFN directly, in regard to broadcasting the frame number. The frame number is conveyed over the PIDS logical channel in Layer 1 as part of a SIS message.

In all AM and FM service modes, the relevant portion of the ALFN being sent applies to the actual frame number at the time it is broadcast. Thus, Layer 1 must ensure proper synchronization of the ALFN being sent relative to absolute GPS time.

5.4 Clock Support

The SIS Transport allows for data to be broadcast making display clocks associated with receivers easier for consumers to use. The provision of these data by a broadcaster is optional if the ALFN is locked to GPS time (Bit 65 of the SIS PDU set to one) and forbidden if it is not (Bit 65 of the SIS PDU set to zero). If present at all, these data may be sent approximately once per minute, or otherwise at the convenience of the broadcaster. Receivers may utilize these data as best suits their design goals.

5.4.1 Handling Leap Seconds

The time standard for clocks around the world is UTC (Coordinated Universal Time). To keep UTC synchronized to astronomical time (defined by the earth's rotation), it is occasionally adjusted by a second. The adjustments average about once a year (so far) and occur as leap seconds, meaning all UTC clocks observe a 61 second minute at midnight when the adjustment occurs. The standard practice is to make adjustments at midnight UTC either December 31 or June 30.

As explained in Subsection 4.3, bit 65 of a PDU must be set to one for the ALFN to be locked to GPS time and for the time of day calculation to be accurate.

HD Radio transmissions may be synchronized to GPS time, which does not have any leap second adjustments. This means that GPS runs ahead of UTC and the receiver derives UTC time as follows:

$$\text{Time(UTC)} = \text{Time(GPS)} - \text{Leap Seconds}$$

$$\text{Time(UTC)} = (65536 / 44100) \cdot \text{ALFN} - \text{Leap Seconds}$$

Between 1980 and 2016, 17 leap seconds have been added to UTC, so GPS is running 17 seconds ahead of UTC. This value is current through December 31st, 2016. On January 1st, 2017, the number of leap seconds will be adjusted to 18. Leap second adjustments may occur periodically; if they occur, they will occur on June 30 or December 31. See Reference [21] for the latest information.

A SIS message is used to convey the current leap second correction factor.

The parameters that are needed to continuously account for leap seconds in the calculation of UTC from the ALFN are:

- the current time offset (GPS time – UTC time),
- ALFN representing the GPS time of a pending leap second adjustment,
- the new time offset after the adjustment.

These parameters are sent over the SIS and should be saved in persistent storage by the receiver so that accurate UTC time can be computed when necessary.

To ensure smooth operation during leap second adjustments, it is suggested that the broadcast system announce leap second adjustments several months in advance. In addition, pending leap second adjustments should continue to be broadcast for at least several hours after the adjustment event has occurred.

Note that the current number of leap seconds as well as the exact time of a pending leap-second adjustment is sent by the GPS satellite constellation as part of the GPS broadcast navigation message. Most GPS-locked time references provide this information and can be used for the HD Radio system.

5.4.2 Handling Local Time

Local time differs from UTC, owing to both the local time zone and the local practice with respect to observing some form of Daylight Saving Time (DST). SIS Parameter Message Index 3 provides digital data on these local customs, so that a receiver's digital display clock can automatically match the local time as spoken in main program audio.

These data describe the local custom at the location of the broadcaster, which may or may not be the same as the local custom at the place of the receiver. Near time-zone boundaries, consumers can receive a multiplicity of stations providing different data. Therefore, these data are provided only as *hints*, the interpretation and utilization of which should be made discretionary, subject to customer control. Receivers may use these data as initial guesses (for example, at initial installation) as to what a persistent configuration should be, with the expectation that consumers may manually adjust the initial guess. (Most of the time, no manual adjustment would be necessary.) Mobile receivers may have a design option to update their clocks with different, localized data as they travel across the country. Or receivers may ignore these data entirely.

AM broadcasters may refrain from transmitting, and AM receivers may refrain from interpreting, these data during evening and nighttime hours.

HD Radio™ FM Transmission

System Specifications

Revision H

December 14th, 2022

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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 details specifications of the iBiquity Digital Corporation HD Radio FM IBOC system. Included in this document are specifications that ensure reliable reception of the digital audio and data, provide precise digital-analog synchronization, define subcarrier power levels, and minimize harmful spectral emissions.

2 Reference Documents

STATEMENT
Each referenced document that is mentioned in this document shall be listed in the following iBiquity document:

- Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard Document Number: SY_REF_2690s

3 Abbreviations, Acronyms, and Conventions

3.1 Abbreviations and Acronyms

AM	Amplitude Modulation
ASF	Secondary Amplitude Scale Factor Select
BPSK	Binary Phase Shift Keying
FCC	Federal Communications Commission
FM	Frequency Modulation
GPS	Global Positioning System
IBOC	In-Band On-Channel
L1	Layer 1
L2	Layer 2
MER	Modulation Error Ratio
MF	Medium Frequency
MP1 – MP3, MP11, MP5, MP6	Standard Primary Service Modes 1 through 3, 11, 5, and 6
MP1X, DSB1	Advanced Primary Service Modes
MP1XOV, MP6OV, DSB1OV	Layered (Hierarchical) Advanced Primary Service Modes
MS5	Secondary Service Mode 5
N/A	Not Applicable
NRSC	National Radio Systems Committee
OBE	Out of Band Emissions
OFDM	Orthogonal Frequency Division Multiplexing
P1	Primary Logical Channel 1
P4	Primary Logical Channel 4
PAR	Peak-to-Average Ratio
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RF	Radio Frequency
SIS	Station Information Service
SSB	Single Side Band
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.
- Bit 0 of a byte or word is considered the least significant bit.
- In representations of binary numbers, the least significant bit is on the right.
- 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.

3.3 Arithmetic Operators

The arithmetic operators used throughout this document are defined below:

Category	Definition	Examples
| x |	Indicates the absolute value of x	| -5 | = 5 | 3 - 4 | = 1

4 FM Transmission Specifications

4.1 Introduction

This document presents the key transmission specifications for the FM HD Radio system.

4.2 Carrier Frequency and Channel Spacing

The HD Radio system operates in-band and on-channel, within the existing allocations and channel spacing as authorized by the FCC in accordance with [12]. The Hybrid and All Digital HD Radio waveforms are centered on the assigned FM band channel frequency.

4.3 Synchronization Tolerances

The system supports two levels of synchronization for broadcasters:

Level I: Network Synchronized (assumed using Global Positioning System (GPS) locked transmission facilities)

Level II: Non-networked Synchronized (non-GPS-locked transmission facilities)

It is recommended that transmission facilities operate as Level I facilities in order to support numerous advanced system features.

4.3.1 Analog Diversity Delay

The absolute accuracy of the analog diversity delay as defined in [1] in the transmission signal shall be within ±68 microseconds (μs) for both Synchronization Level I and Level II transmission facilities. This is equivalent to ±3 audio samples at a sampling rate of 44.1 kHz.

4.3.1.1 Analog Diversity Delay Correction

Correction of the analog diversity delay may be accomplished either in one large step or by smoothly ramping the delay. Under no circumstances shall a delay ramp rate exceed 50 samples per second. When not implementing a correction, the maximum rate of change of the analog diversity delay shall not exceed 3 audio samples (68 μs) per 10 seconds. Refer to Reference [38] for additional guidance on recommended best practices.

4.3.2 Time and Frequency Accuracy and Stability

The total modulation symbol-clock frequency absolute error of an HD Radio broadcast system shall meet the following requirements:

±0.01 ppm maximum for Synchronization Level I facilities

±1.0 ppm maximum for Synchronization Level II facilities

The total digital carrier frequency absolute error shall meet the following requirements:

The total digital carrier frequency absolute error of a Synchronization Level I broadcast system as observed at the RF output shall be ±1.3 Hz maximum.

The total digital carrier frequency absolute error of a Synchronization Level II broadcast system as observed at the RF output shall be ±130 Hz maximum.

4.3.3 Frequency Translators

Frequency translators may be classified as either Synchronization Level I or II regardless of the classification of the primary station. All of the requirements of Subsection 4.3.2 shall apply. In addition, if the translator transmission equipment is operating as Synchronization Level I, and therefore is indicating such condition over the air as part of the SIS data stream, it is strongly recommended that the translator broadcast its own GPS coordinates independently from that of the primary station. This will enhance receiver position-determination capabilities.

4.3.4 On-Channel Boosters

The following requirement shall apply to the use of on-channel boosters:

An on-channel booster shall maintain the same synchronization level as the primary station. All of the requirements of Subsection 4.3.2 shall apply.

In addition, on-channel boosters shall synchronize the content and OFDM symbol timing of their transmissions within ±75 μs relative to the primary station timing at all times, as observed within the coverage area of the booster station. Appropriate delays may be necessary in the studio feed and/or RF transmission path to meet this requirement.

For the purposes of this specification, OFDM symbol timing of 75 μs shall be maintained in the area of mutual interference where booster to main protection ratio is greater than -20 dB. For FM Hybrid transmissions, the booster may be applied to just the digital portion of the Hybrid signal. In this case, the booster antenna to primary antenna field strength ratio shall be computed based on only the digital portion of the signal.

4.3.5 L1 Frame Timing Phase

For Level I transmission facilities, all transmissions shall phase lock their L1 frame timing (and the timing of all OFDM symbols) to absolute GPS time within ±1 μs.

If the above specification in a Synchronization Level I transmission facility is violated, due to a GPS outage or other occurrence, it shall be classified as a Synchronization Level II transmission facility until the above specification is again met.

4.4 FM Spectral Emissions Limits

The requirements for the spectral emissions limits for the Hybrid transmissions and the All Digital transmissions are given in Subsections 4.4.1 and 4.4.2.

4.4.1 Spectral Emissions Limits for Hybrid Transmissions

For Hybrid transmissions, measurements of the combined analog and digital signals shall be made by averaging the power spectral density of the signal in a 1-kHz bandwidth over a minimum time span of 30 seconds and a minimum of 100 sweeps. Compliance will be determined by measuring the composite power spectral density of the analog and digital waveforms. The measurement point and the test configuration shall be as described in Subsection 4.2 of Reference [26].

Zero dBc is defined as the total power of the analog FM carrier.

Under normal operation with analog modulation present, the following requirements shall be met at all times:

Noise and spuriously generated signals from all sources, including phase noise and intermodulation products, shall conform to the limits as described in the following paragraph and shown in Figure 4-1 and Table 4-1. These limits are applicable for all permissible power levels of the upper and lower sidebands, as defined in Subsection 4.5.

The measured power spectral density of the Hybrid analog and digital signals at frequencies removed from the center of the channel between 100 kHz and 200 kHz shall not exceed -30.0 dBc/kHz.

The measured power spectral density of the Hybrid analog and digital signals at frequencies removed from the center of the channel by 200 to 207.5 kHz shall not exceed [-30.0 - (|frequency in kHz| - 200 kHz) · 4.187] dBc/kHz.

The measured power spectral density of the Hybrid analog and digital signals at frequencies removed from the center of the channel by 207.5 to 250 kHz shall not exceed [-61.4 - (|frequency in kHz| - 207.5 kHz) · 0.306] dBc/kHz.

The measured power spectral density of the Hybrid analog and digital signals at frequencies removed from the center of the channel between 250 kHz and 540 kHz shall not exceed -74.4 dBc/kHz.

The measured power spectral density at frequencies removed from the center of the channel by more than 540 to 600 kHz shall not exceed [-74.4 - (|frequency in kHz| - 540 kHz) · 0.093] dBc/ kHz.

The measured power spectral density at frequencies greater than 600 kHz from the center of the channel shall not exceed -80.0 dBc/kHz.

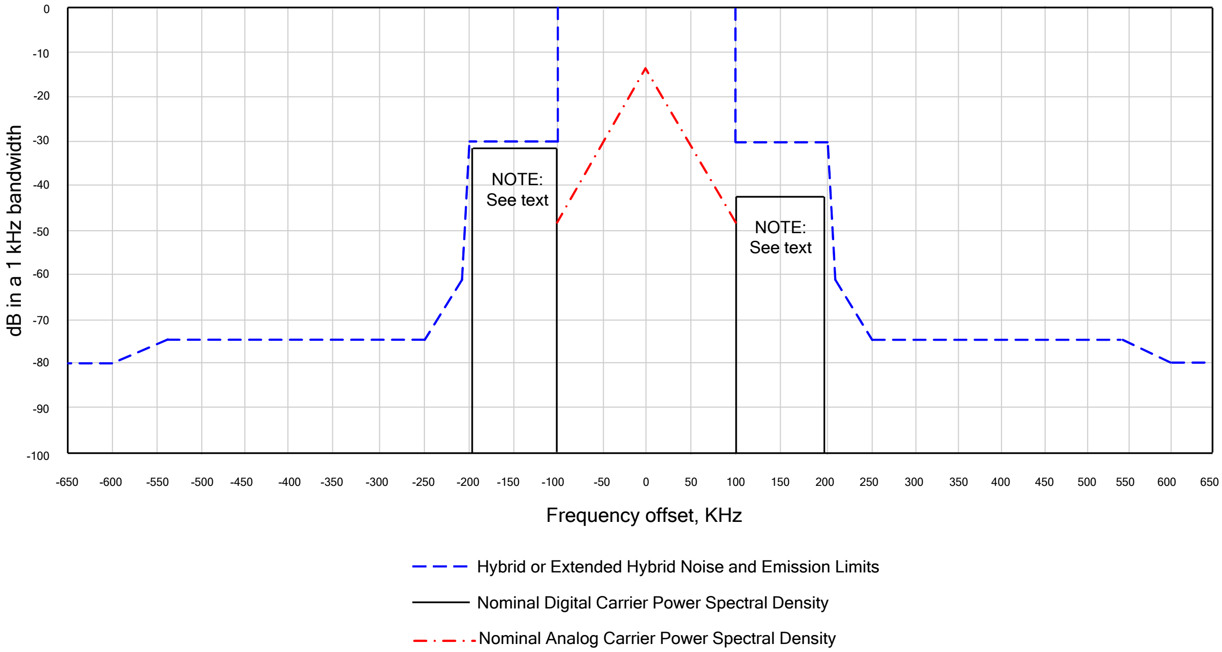

Figure 4-1: HD Radio FM Hybrid Waveform Noise and Emissions Limits

NOTE: The upper and lower sidebands may differ in power level by up to 10 dB (asymmetric sidebands). Normally, the sideband power levels are equal, but under certain scenarios, asymmetric sidebands may be useful for mitigation of adjacent channel interference. Figure 4-1 shows a power-level difference of 10 dB for purposes of illustration. It shall be noted that even though the upper and lower sidebands have different power levels, the upper and lower spectral emissions limits are the same.

Table 4-1: HD Radio FM Hybrid Waveform Noise and Emissions Limits*

Frequency Offset Relative to Carrier	Level, dBc/kHz
100 – 200 kHz offset	-30.0
200 – 207.5 kHz offset	[-30.0 - (|frequency in kHz| - 200 kHz) • 4.187]
207.5 – 250 kHz offset	[-61.4 - (|frequency in kHz| - 207.5 kHz) • 0.306]
250 – 540 kHz offset	-74.4
540 – 600 kHz offset	[-74.4 - (|frequency in kHz| - 540 kHz) • 0.093]
>600 kHz offset	-80.0

* The requirements for noise and spurious emission limits defined in this subsection reflect acceptable performance criteria. In certain circumstances, additional measures (filtering, active emissions suppression, etc.) may be needed to reduce the spectral emissions below the limits given in this subsection in order to reduce mutual interference between broadcast stations.

4.4.2 Spectral Emissions Limits for All Digital Transmissions

For All Digital transmissions, measurements of the All Digital signal shall be made by averaging the power spectral density of the signal in a 1-kHz bandwidth over a minimum time span of 30 seconds and a minimum of 100 sweeps. The measurement point and the test configuration shall be as described in Reference [26].

Zero dBc is defined as the combined total digital power of the upper and lower primary digital sidebands when set to the maximum power spectral density defined in Table 4-3.

Under normal operation, the following requirements shall be met at all times:

Noise and spuriously generated signals from all sources including phase noise and intermodulation products, shall conform to the limits as described in the following paragraph and as shown in Figure 4-2 and Table 4-2‡.

The measured power spectral density of the All Digital signal at frequencies removed from the center of the channel by 200 to 207.5 kHz shall not exceed [-20 - (|frequency in kHz| - 200 kHz) · 1.733] dBc/kHz.

The measured power spectral density at frequencies removed from the center of the channel by more than 207.5 kHz to 250 kHz shall not exceed [-33 - (|frequency in kHz| - 207.5 kHz) · 0.2118] dBc/kHz.

The measured power spectral density at frequencies removed from the center of the channel by 250 to 300 kHz shall not exceed [-42 - (|frequency in kHz| - 250 kHz) · 0.56] dBc/kHz.

The measured power spectral density at frequencies removed from the center of the channel by more than 300 kHz and up to 600 kHz shall not exceed -70 dBc/kHz.

Any emission appearing on a frequency removed from the center of the channel by more than 600 kHz shall not exceed -80 dBc/kHz.

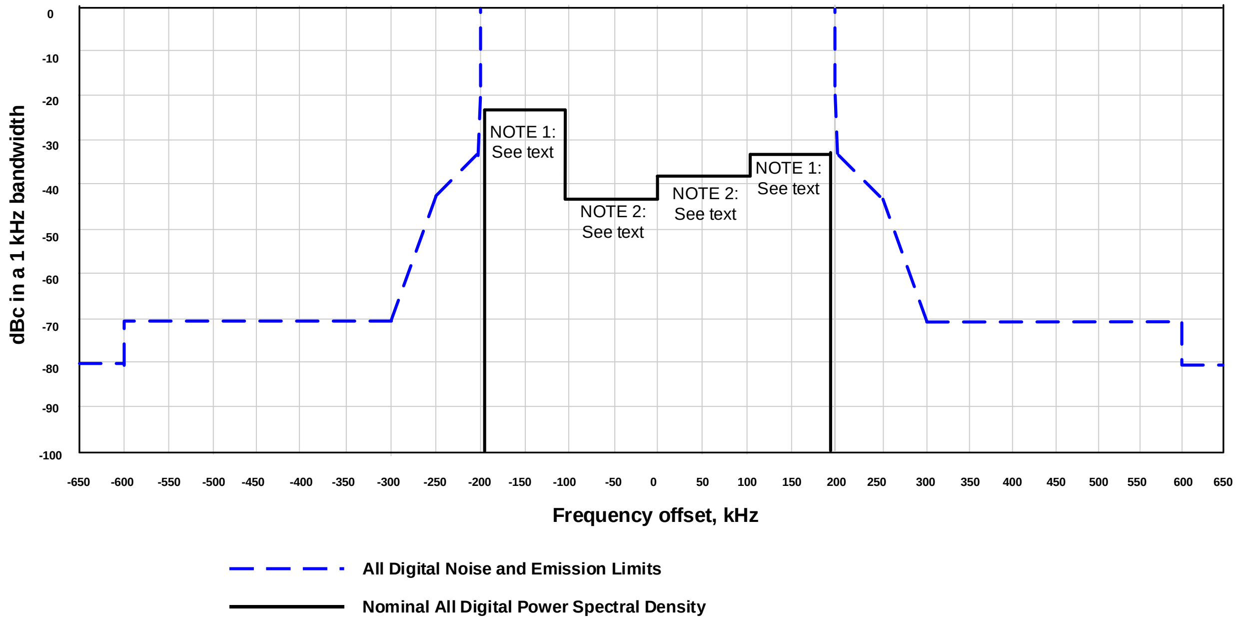

Figure 4-2: HD Radio FM All Digital Waveform Noise and Emissions Limits

NOTE 1: The upper and lower primary sidebands may differ in power level by up to 10 dB (asymmetric sidebands). Normally, the primary sideband power levels are equal, but under certain scenarios, asymmetric sidebands may be useful for mitigation of adjacent channel interference. Figure 4-2 shows a primary sideband power-level difference of 10 dB for purposes of illustration. It shall be noted that even though the upper and lower sidebands have different power levels, the upper and lower spectral emissions limits are the same.

Note 2: For All Digital transmissions, the region within 100 kHz from the center channel is reserved for low-level secondary subcarriers. Like the primary sidebands, the upper and lower secondary sidebands may also differ in power level (asymmetric sidebands). Normally, the secondary sideband power levels are equal, but under certain scenarios, asymmetric sidebands may be useful for mitigation of adjacent channel interference. Figure 4-2 shows a secondary sideband power-level difference of 5 dB for purposes of illustration. In this illustration, the lower secondary sideband is 20 dB below the lower primary sideband, and the upper secondary sideband is 5 dB below the upper primary sideband. Note that broadcasting secondary sidebands is optional in All Digital waveforms.

Table 4-2: HD Radio FM All Digital Waveform Noise and Emissions Limits‡

Frequency Offset Relative to Carrier	Level, dBc/kHz
200 – 207.5 kHz offset	[-20 - (|frequency in kHz| - 200 kHz) • 1.733]
207.5 – 250 kHz offset	[-33 - (|frequency in kHz| - 207.5 kHz) • 0.2118]
250 – 300 kHz offset	[-42 - (|frequency in kHz| - 250 kHz) • 0.56]
300 – 600 kHz offset	-70
>600 kHz offset	-80

‡The requirements for noise and spurious emission limits defined in this subsection reflect acceptable performance criteria. In certain circumstances, additional measures (filtering, active emissions suppression, etc.) may be needed to reduce the spectral emissions below the limits given in this subsection in order to reduce mutual interference between broadcast stations.

4.5 Digital Sideband Levels

The amplitude scaling of each OFDM subcarrier within each digital sideband is given in Table 4-3 for the Hybrid, Extended Hybrid, and All Digital waveforms. The values for the Hybrid and Extended Hybrid waveforms are specified relative to the analog FM power. A value of 1 (0 dBc) would produce a digital subcarrier power equal to the total power in the unmodulated analog FM carrier. The values for the All Digital waveforms are relative to 0 dBc as defined in Subsection 4.4.2, where 0 dBc is the maximum possible combined power level of the primary digital sidebands that could be allocated to any broadcast facility. Therefore, the total authorized digital power in the primary digital sidebands that is allocated to the broadcast facility may be less than 0 dBc for All Digital waveforms with lower-level primary digital sidebands or asymmetric sidebands.

For the Hybrid and Extended Hybrid waveforms, the minimum values of a_{0U} and a_{0L} were chosen so that the total average power in a primary *main* digital sideband (upper or lower) is 23 dB below the total power in the unmodulated analog FM carrier. The power of each primary sideband may be individually increased according to the maximum values shown in Table 4-3. Therefore, total average power in each primary main digital sideband (upper or lower) is subject to an upper limit of 13 dB below the total power in the unmodulated analog FM carrier. For Extended Hybrid waveforms, the total average power in each primary digital sideband is subject to different upper and lower limits, according to primary service mode, as shown in Table 4-4. Normally, the upper and lower sideband power levels are equal, but under certain scenarios, asymmetric sidebands may be useful for mitigation of adjacent channel interference.

For the All Digital waveform, the maximum values of a_{1U} and a_{1L} were chosen so that the total average power of all the primary digital subcarriers in both sidebands combined would be equal to one. The power of each primary sideband may be individually decreased according to the minimum values shown in Table 4-3. Therefore, total average power in each primary digital sideband (upper or lower) is subject to an upper limit of -3 dB and a lower limit of -13 dB below the total authorized digital power. Normally, the upper and lower primary sideband power levels are equal, but under certain scenarios, asymmetric sidebands may be useful for mitigation of adjacent channel interference.

One of four All Digital secondary sideband power levels may be independently applied to the upper and lower secondary sidebands, resulting in asymmetric sidebands for mitigation of adjacent channel interference. Minimum and maximum values for a_{2L} through a_{5L} and a_{2U} through a_{5U} were chosen so that the total average power in the upper or lower secondary digital sideband lies in the range of 5 to 20 dB below the total power in the upper or lower primary digital sideband, respectively. The selection of one of the values a_{2L} through a_{5L} and one of the values a_{2U} through a_{5U} is determined by the secondary amplitude scale factor select (ASF) received from L2.

Table 4-3: OFDM Subcarrier Scaling

Waveform	Service Mode	Sidebands	Amplitude Scale Factor Notation	Power Spectral Density, dBc per Subcarrier		Power Spectral Density in a 1-kHz Bandwidth, dBc	
				Min	Max	Min	Max
Hybrid	MP1	Primary	a_{0L}	-45.8	-35.8	-41.4	-31.4
			a_{0U}	-45.8	-35.8	-41.4	-31.4
Extended Hybrid	MP2, MP3, MP11, MP5, MP6, MP1X, DSB1, MP1XOV, MP6OV, DSB1OV	Primary	a_{0L}	-45.8	-35.8	-41.4	-31.4
			a_{0U}	-45.8	-35.8	-41.4	-31.4
All Digital	MP5, MP6, DSB1, MP6OV, DSB1OV	Primary	a_{1L}	-37.3	-27.3	-32.9	-22.9
			a_{1U}	-37.3	-27.3	-32.9	-22.9
	MS5	Secondary	a_{2L}	-42.3	-32.3	-37.9	-27.9
			a_{2U}	-42.3	-32.3	-37.9	-27.9
			a_{3L}	-47.3	-37.3	-42.9	-32.9
			a_{3U}	-47.3	-37.3	-42.9	-32.9
			a_{4L}	-52.3	-42.3	-47.9	-37.9
			a_{4U}	-52.3	-42.3	-47.9	-37.9
			a_{5L}	-57.3	-47.3	-52.9	-42.9
			a_{5U}	-57.3	-47.3	-52.9	-42.9

NOTE: The values in Table 4-3 represent average power levels. Signal constellations have been adjusted such that all waveforms with different modulation orders (QPSK, 16-QAM, or 64-QAM) will have the same average power. However, peak voltages may increase in advanced service modes that use higher-order modulation techniques, and accommodations may need to be made in component selection and design.

4.5.1 FM Hybrid and Extended Hybrid Digital Carrier Power

4.5.1.1 Hybrid and Extended Hybrid System Carrier Configuration

Hybrid transmission utilizes two OFDM subcarrier sets (sidebands) located up to 198 kHz above and below the analog carrier center frequency. The basic hybrid (MP1) service mode uses 191 subcarriers per sideband beginning in frequency at approximately ±129 kHz from the center frequency. Extended hybrid service modes MP2, MP3, MP11, MP5, MP6, MP1X, DSB1, MP1XOV, MP6OV, and DSB1OV add additional subcarriers closer to the analog carrier, with all but MP2's and MP3's subcarriers starting at approximately ±101 kHz.

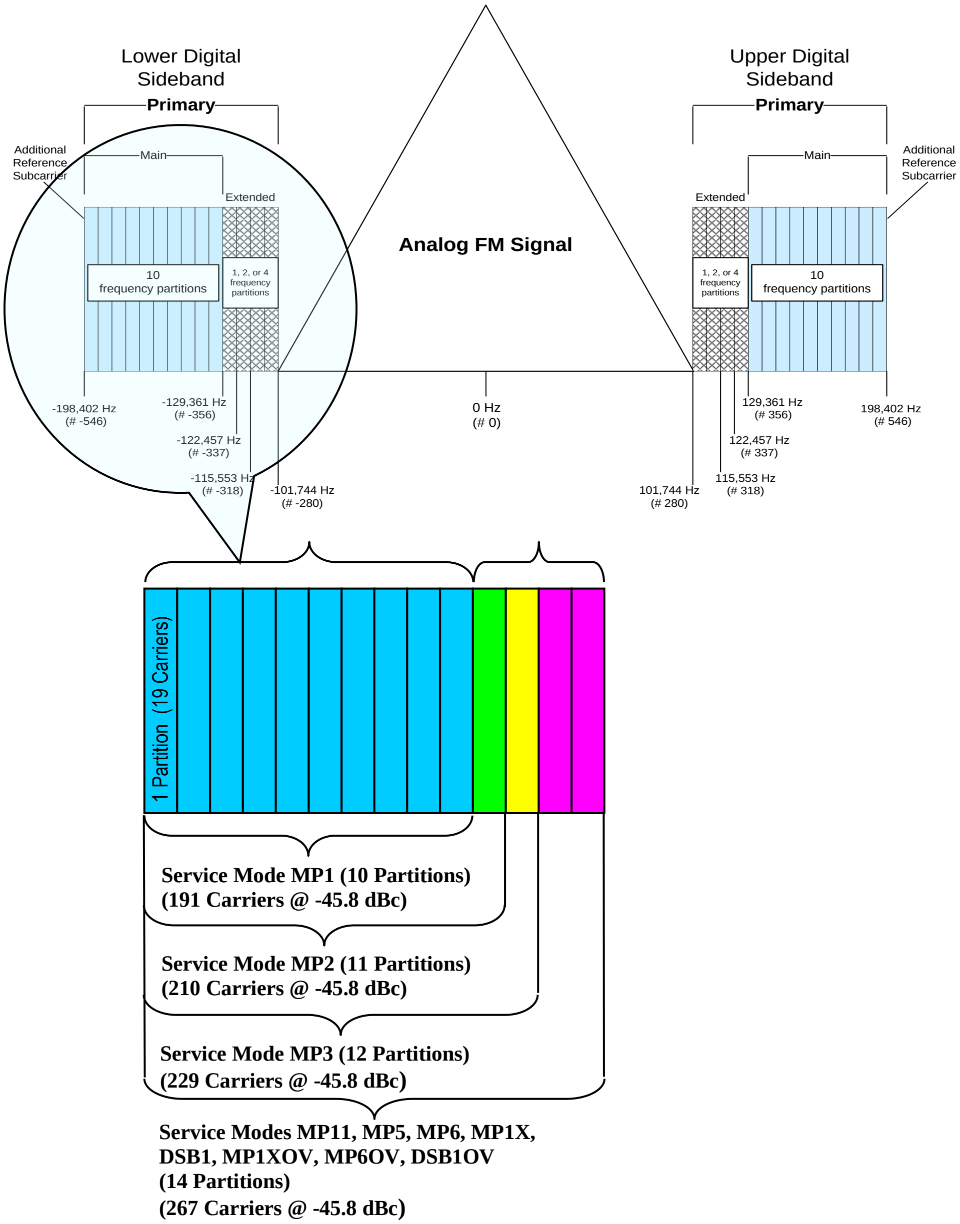

Figure 4-3: Sideband Detail of the Hybrid and Extended Hybrid Waveforms

As illustrated in Figure 4-3, each frequency partition consists of 19 subcarriers (except for two extra reference subcarriers at the limits of the primary main partitions). In the Lower Digital Sideband, Figure 4-3 also details each of the sideband groups for all hybrid and extended hybrid service modes. The power of each subcarrier is set at -45.8 dBc (dB below the reference analog carrier) for a -20-dBc total integrated digital-to-analog power ratio in service mode MP1 (including both upper and lower digital sidebands).

Table 4-4 characterizes power at other digital-to-analog power ratios in FM hybrid and extended hybrid service modes. Since the absolute power of the subcarriers is additive, if 10 subcarrier groups make up the MP1 reference power level, one more group will increase the power by 10% and so on. Note that there is one extra reference subcarrier in each of the primary main sidebands, skewing the power calculation slightly. This amounts to approximately 0.4% of the total power in the MP3 mode or 0.02 dB, which is considered negligible.

4.5.1.2 Digital Power for Hybrid and Extended Hybrid Waveforms at Various Digital-to-Analog Power Ratios

Table 4-4 characterizes the total integrated digital power and single-sideband power for the hybrid and extended hybrid service modes and various digital-to-analog power ratios. The nominal digital-to-analog power ratio is derived from Table 4-3 assuming a digital-to-analog power ratio of -20 dBc. Other power ratios are scaled appropriately as referenced in Table 4-4.

Table 4-4: Sideband Power for Various Service Modes and Digital to Analog Power Ratios

Nominal Digital-to-Analog Power Ratio (dBc) Service Mode MP1	Single Subcarrier Power (dBc)	Total Integrated Power of Both Sidebands (dBc)				Total Integrated Power of One Sideband (dBc)			
		MP1 100% of MP1 Power	MP2 110% of MP1 Power	MP3 120% of MP1 Power	OTHER[1] 140% of MP1 Power	MP1 100% of MP1 Power	MP2 110% of MP1 Power	MP3 120% of MP1 Power	OTHER[1] 140% of MP1 Power
-20.0	-45.8	-20.0	-19.6	-19.2	-18.5	-23.0	-22.6	-22.2	-21.5
-14.0	-39.8	-14.0	-13.6	-13.2	-12.5	-17.0	-16.6	-16.2	-15.5
-13.0	-38.8	-13.0	-12.6	-12.2	-11.5	-16.0	-15.6	-15.2	-14.5
-12.0	-37.8	-12.0	-11.6	-11.2	-10.5	-15.0	-14.6	-14.2	-13.5
-11.0	-36.8	-11.0	-10.6	-10.2	-9.5	-14.0	-13.6	-13.2	-12.5
-10.0	-35.8	-10.0	-9.6	-9.2	-8.5	-13.0	-12.6	-12.2	-11.5

[1]OTHER service modes include MP11, MP5, MP6, MP1X, DSB1, MP1XOV, MP6OV, DSB1OV

4.5.1.3 Power Limits for Asymmetrical Sideband Operation

If asymmetrical sideband operation is desired, the lower and upper digital sidebands are considered separately and the single-sideband power values in Table 4-4 are used. Note that these values are simply three dB less than the corresponding total integrated power for both sidebands. If broadcasting in MP3 mode, for example, setting the lower sideband at -10 dBc and the upper at -14 dBc will result in a total integrated power of:

$$= 10 \text{ Log}_{10} (\text{Log}_{10}{}^{-1} (\text{Pwr1} /10) + \text{Log}_{10}{}^{-1} (\text{Pwr2} /10))$$

$$= 10 \text{ Log}_{10} (\text{Log}_{10}{}^{-1} (-12.2 /10) + \text{Log}_{10}{}^{-1} (-16.2 /10))$$

$$= 10 \text{ Log}_{10} (0.060 + 0.024)$$

$$= 10 \text{ Log}_{10} (0.084)$$

$$= -10.8 \text{ dBc}$$

Note that the total integrated power is dominated by the highest-powered sideband.

4.5.2 RF Spectral Inversion

The RF spectrum of the digital waveform shall be inverted as compared to its baseband representation. This means that the lower sideband shall occupy the higher frequencies within the RF channel. And the upper sideband shall occupy the lower frequencies within the RF channel. Hence, scale factor a_{0L} shall be used to set the power level of the higher frequency sideband and a_{0U} shall be used to set the power level of the lower frequency sideband.

Refer to Subsection 14.2.2 of [1] for further details.

4.6 Phase Noise

The phase noise mask for the broadcast system is illustrated in Figure 4-4 and Figure 4-5 and specified in Table 4-5. As can be seen in the figures, the response is linear (on the dB scale) between every pair of points drawn on the curve.

Phase noise is inclusive of all sources from the Exciter input to the antenna output as measured in a 1-Hz bandwidth.

Zero dBc is defined as the total power of the subcarrier being measured. The phase noise mask is applicable for all permissible power levels of the upper and lower sidebands, as defined in Subsection 4.5.

The total single sideband phase noise of any digital subcarrier at the transmitter RF output as measured in a 1-Hz bandwidth shall be within the mask specified in Table 4-5. This shall be verified by transmitting a single unmodulated digital subcarrier. In addition, for the Hybrid waveform, the analog FM carrier shall be disabled.

Table 4-5: FM Broadcast System Phase Noise Specification

Frequency Offset Relative to Carrier (F)	Level, dBc/Hz
10 Hz to 100 Hz	-2.78×10^{-1} F - 39.2
100 Hz to 1000 Hz	-1.11×10^{-2} F - 65.9
1 kHz to 10 kHz	-1.11×10^{-3} F - 75.9
10 kHz to 100 kHz	-2.22×10^{-4} F - 84.8
> 100 kHz	-107.0

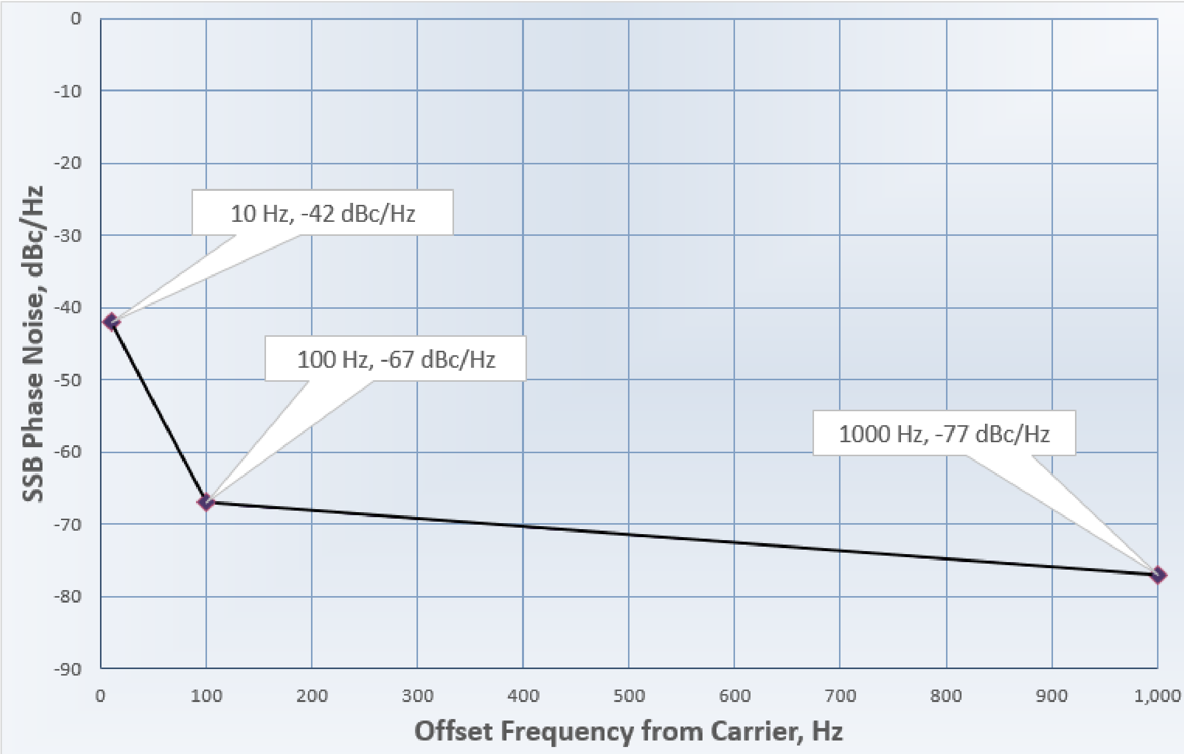

Figure 4-4: FM SSB Phase Noise Mask | 10 Hz to 1000 Hz

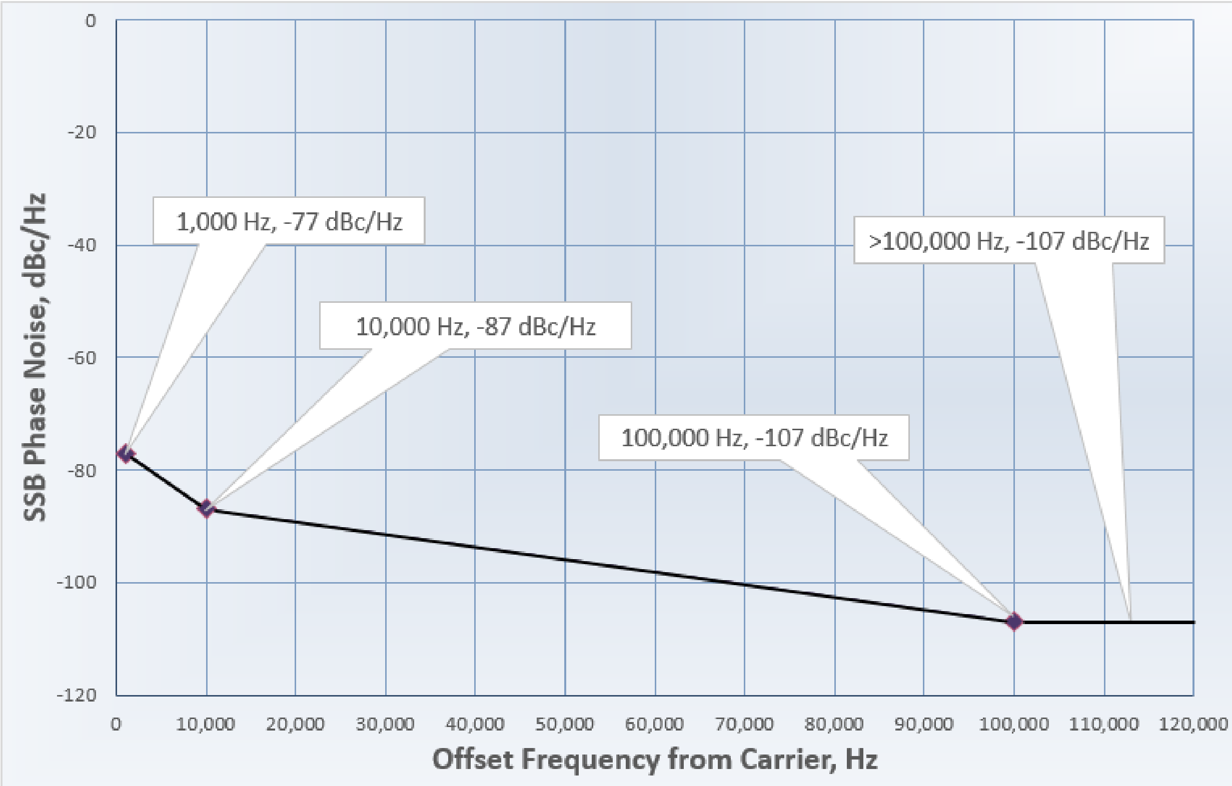

Figure 4-5: FM SSB Phase Noise Mask | 1 kHz to 100 kHz

4.7 Discrete Phase Noise

For the broadcast system, the spectrum from (F_c – 200 kHz) to (F_c + 200 kHz) shall be considered to consist of multiple non-overlapping sub-bands, each with a bandwidth of 300 Hz, where F_c is the carrier frequency. Discrete phase noise components measured at the transmitter RF output shall be permitted to exceed the mask specified in Table 4-5 provided that for each sub-band, the measured total integrated phase noise does not exceed the total integrated phase noise calculated from Table 4-5.

If the upper and lower sidebands have different power levels, as permitted in Subsection 4.5, the measurement must account for the fact that the 0-dBc reference level will be different for each sideband.

4.8 Modulation Error Ratio

Modulation Error Ratio (MER) is a useful signal quality metric, quantifying the ratio of the rms noise of one or more subcarriers to the subcarrier nominal magnitude(s). Thus, it is a measure of the signal-to-noise ratio (in units of dB) of the broadcast signal, inclusive of both linear and non-linear distortions within the broadcast system itself. Refer to Reference [27] for details of how MER is measured and computed.

The following specifications shall be met, using the test configuration described in Subsection 4.2 of Reference [26].

4.8.1 Reference Subcarriers

1. The MER for each and every Binary Phase Shift Keying (BPSK) reference subcarrier, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 11 dB, as computed by Equation 1 in Reference [27]. The parameter N in Equation 1, the total number of contiguous symbols used in the average, shall be set to 128.
2. The average MER of all the BPSK reference subcarriers in the upper primary sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 14 dB, averaged across all upper primary reference subcarriers, as computed by Equation 2(a) in Reference [27]. This computation shall be based on a block of N = 128 contiguous symbols.
3. The average MER of all the BPSK reference subcarriers in the lower primary sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 14 dB, averaged across all lower primary reference subcarriers, as computed by Equation 2(b) in Reference [27]. This computation shall be based on a block of N = 128 contiguous symbols.
4. The average MER of all the BPSK reference subcarriers in the upper secondary sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 14 dB, averaged across all upper secondary reference subcarriers, as computed by Equation 2(c) in Reference [27]. This computation shall be based on a block of N = 128 contiguous symbols.
5. The average MER of all the BPSK reference subcarriers in the lower secondary sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 14 dB, averaged across all lower secondary reference subcarriers, as computed by Equation 2(d) in Reference [27]. This computation shall be based on a block of N = 128 contiguous symbols.

4.8.2 Data Subcarriers for Standard Service Modes (MP1, MP2, MP3, MP5, MP6, and MP11)

1. The MER for each and every data-subcarrier partition, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 11 dB, as computed by Equation 4 in Reference [27]. In Equation 4, the noise voltage $Vndat_m$ is calculated for Quadrature Phase Shift Keying (QPSK) modulation using the sideband (upper primary or lower primary) where the partition is located. The parameter N, the total number of contiguous symbols used in the calculation of $Vndat_m$, shall be set to 128.
2. The average MER of all QPSK data subcarriers in the upper primary sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 14 dB, averaged across all upper primary data subcarrier partitions, as computed by Equation 5(a) in Reference [27]. This computation shall be based on a block of N = 128 contiguous symbols.
3. The average MER of all QPSK data subcarriers in the lower primary sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 14 dB, averaged across all lower primary data subcarrier partitions, as computed by Equation 5(b) in Reference [27]. This computation shall be based on a block of N = 128 contiguous symbols.

4.8.3 Data Subcarriers for Advanced Service Modes (MP1X, DSB1, MP1XOV, MP6OV, DSB1OV, and MS5)

Advanced service modes are comprised of quadrature amplitude modulation (QAM) data subcarriers, each of which uses one of three possible modulation orders: QPSK (4-QAM), 16-QAM, or 64-QAM. In some advanced service modes, more than one modulation order may be employed across an upper or lower primary sideband. For example, service mode MP1X uses QPSK for the P1 logical channel on the primary main frequency partitions, and 16-QAM for the P4 logical channel on the primary extended frequency partitions. Therefore, MER values for advanced service modes are specified for each applicable modulation order over its assigned frequency partitions. See Table 4-6.

1. The MER for each and every data-subcarrier partition, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to the value specified in Table 4-6, as computed by Equation 4 in Reference [27]. In Equation 4, the noise voltage $Vndat_m$ is calculated for QPSK, 16-QAM, or 64-QAM modulation using the sideband (upper primary, lower primary, upper secondary, or lower secondary) where the partition is located. The parameter N, the total number of contiguous symbols used in the calculation of $Vndat_m$, shall be set to 128.
2. The average MER of all QPSK data subcarriers in the upper primary sideband for the specified advanced service mode, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than the value specified in Table 4-6, averaged across all QPSK-modulated upper-primary-sideband data-subcarrier partitions, as computed by Equation 5(a) in Reference [27]. This computation shall be based on a block of N = 128 contiguous symbols.
3. The average MER of all QPSK data subcarriers in the lower primary sideband for the specified advanced service mode, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than the value specified in Table 4-6, averaged across all QPSK-modulated lower-primary-sideband data-subcarrier partitions, as computed by Equation 5(b) in Reference [27]. This computation shall be based on a block of N = 128 contiguous symbols.
4. The average MER of all 16-QAM or 64-QAM data subcarriers in the upper primary sideband for the specified advanced service mode, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than the value specified in Table 4-6, averaged across all 16-QAM-modulated or 64-QAM-modulated upper-primary-sideband data-subcarrier partitions, as computed by Equation 5(c) in Reference [27]. This computation shall be based on a block of N = 128 contiguous symbols.
5. The average MER of all 16-QAM or 64-QAM data subcarriers in the lower primary sideband for the specified advanced service mode, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than the value specified in Table 4-6, averaged across all 16-QAM-modulated or 64-QAM-modulated lower-primary-sideband data-subcarrier partitions, as computed by Equation 5(d) in Reference [27]. This computation shall be based on a block of N = 128 contiguous symbols.
6. The average MER of all QPSK data subcarriers in the upper secondary sideband for advanced service mode MS5, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than the value

specified in Table 4-6, averaged across all QPSK-modulated upper-secondary-sideband data-subcarrier partitions, as computed by Equation 5(e) in Reference [27]. This computation shall be based on a block of N = 128 contiguous symbols.

7. The average MER of all QPSK data subcarriers in the lower secondary sideband for advanced service mode MS5, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than the value specified in Table 4-6, averaged across all QPSK-modulated lower-secondary-sideband data-subcarrier partitions, as computed by Equation 5(f) in Reference [27]. This computation shall be based on a block of N = 128 contiguous symbols.

Table 4-6: MER Specifications for Advanced Service Modes

Advanced Service Mode	**Modulation Order**	**Frequency Partitions**	**Average MER per Sideband, dB**	**Minimum MER per Individual Partition, dB**
MP1X	QPSK	Primary Main	14	11
	16-QAM	Primary Extended	19	16
DSB1	16-QAM	Primary Main + Primary Extended	17	14
MP1XOV	16-QAM	Primary Main + Primary Extended	19	16
MP6OV	16-QAM	Primary Main + Primary Extended	17	14
DSB1OV	64-QAM	Primary Main + Primary Extended	25	22
MS5	QPSK	Secondary	20	17

Manufacturers of IBOC (HD Radio)-compatible transmitters and amplifiers shall consider the effects of service mode, with and without peak-to-average ratio (PAR) reduction, on MER performance and service rating. In particular, with the transmission of new advanced FM service modes, broadcast equipment manufacturers must consider increasing voltage safety margins and associated peak power levels across the range of potential HD Radio signal types when developing their products.

Appendix C of Reference [27] provides simulation results indicating peak power levels by service mode to aid manufacturers in selecting components and minimum voltage safety factors on transmission components.

All-digital advanced FM service modes with higher-order modulation produce the highest expected signal peaks. As the analog host signal is no longer present in the all-digital waveform, the superposition of OFDM digital subcarriers dominates output signal characteristics. If the transmitter design employs PAR reduction, peak power levels are effectively reduced.

4.8.4 Data Subcarrier to Reference Subcarrier Power Ratio

In addition to the gain flatness specifications stated in Subsection 4.9, the ratio of the average data subcarrier power to the average reference subcarrier power shall comply with the following limits:

$$-0.5 \le RdB \le 1.0$$

as computed by the following equations defined in Section 12 of Reference [27]:

Equation 3(a) for QPSK data subcarriers in the upper primary sideband;

Equation 3(b) for QPSK data subcarriers in the lower primary sideband;

Equation 3(c) for 16-QAM or 64-QAM data subcarriers in the upper primary sideband;

Equation 3(d) for 16-QAM or 64-QAM data subcarriers in the lower primary sideband;

Equation 3(e) for QPSK data subcarriers in the upper secondary sideband;

Equation 3(f) for QPSK data subcarriers in the lower secondary sideband.

4.9 Gain Flatness

The total gain of the transmission signal path as verified at the antenna output shall be flat to within ±0.5 dB for all frequencies between (F_c – 200 kHz) to (F_c + 200 kHz), where F_c is the RF channel frequency. It is assumed that the source data consists of scrambled binary ones and the power of each subcarrier is an average value.

For the case where the upper and lower digital sideband power levels are intended to be different, as defined in Subsection 4.5, the gain flatness specification shall be interpreted as follows:

> Gain flatness is the difference between the measured power spectral density in a 1-kHz bandwidth of each subcarrier frequency, and the power spectral density of the applicable digital Primary Main sideband, normalized to a 1-kHz bandwidth.

For optimal HD Radio digital performance, it is recommended that the transmission system, including the antenna, adheres as closely as is practicable to the Gain Flatness specification. Performance may be verified using a suitable sample loop on the reference or main tower. In addition to antenna component selection and adjustment, active pre-compensation of the HD Radio waveform may be employed to improve the effective gain flatness.

4.10 Group Delay Flatness

The differential group delay variation of the entire transmission signal path (excluding the RF channel) as measured at the RF channel frequency (F_c) shall be within 900 ns peak to peak from (F_c – 200 kHz) to (F_c + 200 kHz).

HD Radio™ Air Interface Design Description Program Service Data

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SY_IDD_1028s

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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 provides a description of the Program Service Data capability. Program Service Data refers to both Main Program Service Data (MPSD) and Supplemental Program Service Data (SPSD). A detailed description of ID3v2.3.0 message encoding is explained in Appendix A. Appendix A contains the ID3 standard as implemented in the HD Radio system and the relevant sections of the specifications.

Refer to Reference [NRSC-G200-A] for guidelines on harmonizing PSD data fields with RBDS/RDS transmissions.

2 Referenced Documents

STATEMENT
Each referenced document that is mentioned in this document shall be listed in the following iBiquity document:

- Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard Document Number: SY_REF_2690s

3 Abbreviations, Acronyms, and Conventions

3.1 Abbreviations and Acronyms

AAS	Advanced Application Services
AM	Amplitude Modulation
API	Application Programming Interface
CODEC	Coder Decoder
COMM	Comment (ID3 Frame ID)
COMR	Commercial (ID3 Frame ID)
FM	Frequency Modulation
HD RLS	HD Radio Link Subsystem
IBOC	In-Band On-Channel
ID	Identification
MF	Medium Frequency
MPEG	Motion Picture Experts Group
MPS	Main Program Service
MPSA	Main Program Service Audio
MPSD	Main Program Service Data
PDU	Protocol Data Unit
PSD	Program Service Data
REFID	Reference Identifier
SIS	Station Information Service
SPS	Supplemental Program Service
SPSA	Supplemental Program Service Audio
SPSD	Supplemental Program Service Data
TALB	Album (ID3 Frame ID)
TCON	Genre (ID3 Frame ID)
TIT2	Title (ID3 Frame ID)
TPE1	Artist (ID3 Frame ID)
UFID	Unique File Identifier (ID3 Frame ID)
URL	Uniform Resource Locator
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.

4 Overview

The Main Program Service (MPS) allows the transmission of existing analog radio programming in both analog and digital formats. MPS includes the Main Program Service Audio (MPSA) and the Main Program Service Data (MPSD). Similarly, the Supplemental Program Service Data includes the Supplemental Program Service Audio (SPSA) and the Supplemental Program Service Data (SPSD). Both MPSD and SPSD are generally referred to as Program Service Data (PSD). PSD provides additional information about the audio. The processing of the Supplemental Program Service Data (SPS Data) is exactly the same as the MPS Data.

The main program audio and data are synchronized at the broadcast studio. That is, the PSD is transmitted so that radio receivers can acquire it and present it while its associated audio program is being heard by radio listeners.

4.1 Program Service Data Flow

Figure 4-1 illustrates the PSD flow through the HD Radio Broadcast System. The PSD is sent through the Service Interface as ID3 tags which are transmitted as data packets. The PSD is processed by the HD RLS mechanism within the PSD Transport [10] for framing and encapsulation as byte streams and then inserted into the audio streams by the Audio Transport as PSD PDUs (MPSD/SPSD PDUs). The Audio Transport [4] then multiplexes the PSD PDUs with the audio streams to generate MPS/SPS PDUs.

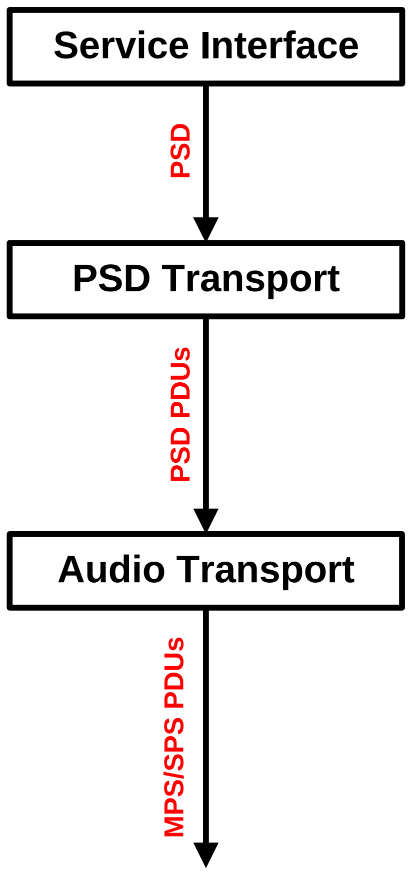

Figure 4-1: Program Service Data Flow

5 Program Service Data Description

Program Service Data (PSD) is transmitted along with the program audio. PSD is intended to describe or complement the audio program heard by the radio listener.

The following subsections provide:

- An introduction to the basic PSD content
- A description of broadcast PSD processing
- The format of PSD messages

5.1 Basic PSD Content

PSD consists of a general set of categories that describe the various programming content, such as a song, talk show, advertisement, or announcement. For example, the Title field can be used to describe the name of a song, topic of a talk show, advertisement, or announcement.

The PSD fields include the following:

- Title
- Artist
- Album
- Genre
- Comment
- Commercial
- Reference Identifier

A detailed description for the format of PSD messages is discussed in Subsection 5.3.

5.2 Broadcast PSD Processing

PSD can originate from a studio automation system or any other computing resource where program audio originates. Regardless of the source, the processing and interface to facilitate broadcast of PSD is consistent. Program Service Data providers input the desired content (for example, title, artist, album, etc.) and transfer the resulting PSD message to the Service Interface. The PSD data packets then undergo framing and encapsulation by the HD RLS as byte streams within the PSD Transport before being sent as PSD PDUs to the Audio Transport where they are multiplexed with the audio streams.

Some key considerations for Program Service Data are:

- Typical PSD messages contain less than 30 characters per field.
- PSD messages are continuously transmitted, with the most recent message transmitted repeatedly.
- PSD providers send a new PSD message only when the PSD content has changed. Note that radio receivers will continue to display information for a given ID3 frame until one of the following things occur:
    - A new ID3 tag is received with that particular frame omitted. In this case, the receiver shall clear the text for that frame.
    - A new ID3 tag is received with different text in that particular frame. In this case, the receiver shall display the updated text for that frame.
    - The signal is lost for an appreciable amount of time. In this case, all ID3 tag information shall be cleared by the receiver.

5.3 Format of PSD Messages

PSD is formatted using a subset of the standard called ID3v2.3.0 (Reference [25]). Historically, ID3 has been used to allow textual information, such as artist, title, and genre information, to co-exist within MPEG-3 (MP3) audio files. The HD Radio system uses ID3 to deliver Program Service Data along with real-time broadcast audio.

The HD Radio system implements a specific subset of the ID3v2.3.0 parameters. The ID3v2.3.0 general structure is as follows:

- A complete ID3 message is called an ID3 tag.
- ID3 tags contain one or more content types referred to as frames. Frames contain individual pieces of information (for example, song, artist, title, etc.). Each frame has a four-character identifier. For example, the commercial frame is identified as COMR.
- Within frames, sub-elements called fields can exist. Fields further categorize the information within a frame. For example, the commercial frame has a field to specify sale price.

Figure 5-1 shows the general ID3 message structure.

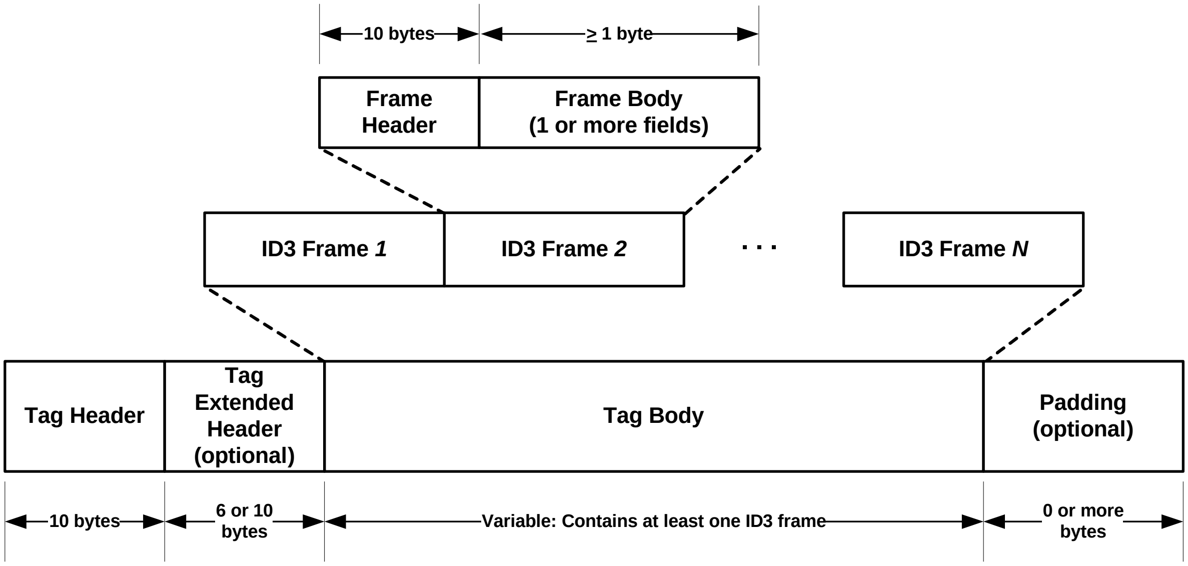

Figure 5-1: General ID3 Message Structure

Appendix A contains the ID3 standard as implemented in the HD Radio system and the relevant sections of the specification.

Table 5-1 gives a description of ID3 frames used for PSD. For a detailed description of ID3v2.3.0 message encoding used for PSD, see Appendix A.

Table 5-1: ID3 Frames Supported by PSD

	PSD	ID3	ID3	Description	Type		
	Attribute	Frame ID	Field(s)		Music	Talk	Announcement
1	Title	TIT2	Info	One-line Title Name	Song title	Talk Topic	Announcement or Advertisement Title
2	Artist	TPE1	Info	Performer, Originator, Author, Sponsor	Artist Name	Show Host	Author/Sponsor
3	Album	TALB	Info	Content Source	Album Name	Show Name	Sponsor Name
4	Genre	TCON	Info	Categorization of content This is an enumerated field of predefined types	(8) Jazz (17) Rock (32) Classical	(101) Speech	(101) Speech
5	Comment	COMM	Language	3-byte language code per ISO 639-2	3 byte code	3-byte code	3-byte code
			Short description field	One-line Title for Comment Description	Comment Title (or complete comment if Content field is omitted)	Comment Title (or complete comment if Content field is omitted)	Comment Title (or complete comment if Content field is omitted)

	PSD	ID3	ID3	Description	Type		
	Attribute	**Frame ID**	**Field(s)**		**Music**	**Talk**	**Announcement**
5	Comment	COMM	Content field	Comment Description Detailed explanation, user callback information or further information	Further information about the song such as recording date, artist bio info. Announcements of general interest such as the weather, concert dates.	Talk Show call-in number, or other show info Announcements of general interest such as the weather, concert dates.	Announcement or Advertising statement Point-of-sale or more info Announcements of general interest such as the weather, concert dates.
6	Commercial	COMR	Price	Price of merchandise	The commercial frame facilitates sale of products and services (Note: Binary pictures are not supported. Any images shall be sent separately via the AAS or LOT data transports.)		
			Valid until	Expiration data for transaction			
			Contact URL	URL identifier used to contact the seller Can be used to initiate purchase transaction via an external return channel, such as a cellular phone network			
			Received as	Method in which merchandise is received (e.g., over the internet)			
6	Commercial	COMR	Name of seller	Text identifying seller			
			Description	Textual description of advertisement			
			Picture	Picture of advertised item This item is not supported and shall not be broadcast.			
			Seller Logo	Binary graphic of seller logo This item is not supported and shall not be broadcast.			

	PSD	ID3	ID3		Type		
	Attribute	**Frame ID**	**Field(s)**	**Description**	**Music**	**Talk**	**Announcement**
7	Reference Identifier	UFID	Owner Identifier	Typically a URL that may be used to connect with the audio content in some way.	---		
			Identifier	Unique Identifier	An identifier code, in either binary or text format of up to 64 bytes in length.		

6 Requirements for Program Service Data

Some key requirements for Program Service Data are:

- Program audio and associated data shall be synchronized as tightly as possible so that receivers will render them both at the same time.
- PSD messages shall not exceed 1024 bytes, including HD RLS overhead as described in [5]. Therefore, ID3 tags shall be limited to no larger than 1,018 bytes.
- PSD shall utilize only the subset of the ID3v2.3.0 standard as shown in Table 5-1.
- PSD is not compatible with later versions of ID3; such as v2.4.0.
- Broadcasters providing PSD shall, at a minimum, transmit the Title and Artist information.
    - If dynamic program Info is not available, only the Song Title field shall be populated with static information. The Artist field shall be left empty.
- Title, Artist, Album, and Genre frames shall be limited to less than 128 characters (excluding the frame header).
- Comment, Commercial, and Reference Identifier frames shall be any length as long as the maximum ID3 tag size of 1,018 characters is not exceeded.
- ID3 frames shall not contain empty strings or null strings (single 0x00 byte) after the frame header. Frames should not be sent unless they contain at least one displayable character.
- The text encoding byte that is specified in the ID3 specification is required for all text-based frames. A value of 0x00 indicates ISO-8859-1 encoding and is highly recommended if possible. A value of 0x01 indicates Unicode, which is a much richer character set, but may not be supported by all receivers.
- All ID3 tags shall start with the characters "ID3" followed by version bytes of 0x03 0x00. Other versions are not supported.

7 ID3 Tag Implementation Considerations

This section provides operational constraints that should be applied to ID3 tag generation in order to ensure successful ID3 tag decoding and display by radio receivers.

The following features defined by the ID3 specification are not supported by the HD Radio system. Refer to Appendix A for details.

- Padding bytes are not supported
- Unsychronization is not supported. ID3 tags are always byte-aligned by the PSD data transport mechanism. No furteh rsynchronization is needed.
- CRC is not supported. The PSD data transport already includes its own CRC capability.
- Data compression is not supported. The maximum tag and frame size constraints stated in Section 6 shall always apply. In addition, it is suggested to keep ID3 tags and frames well below these limits in order to minimize transmission latency. For example, reference [G-200A] provides recommendations on reduced frame sizes in order to maintain compatibility with equivalent frames sent via RBDS.
- Data encryption at the ID3 frame or tag level is not supported.

7.1 Genre Frames

Genre should be considered non-displayable by radio receivers. The genre frame is intended to identify the specific genre of each individual song. The genre codes, as defined by the ID3 standard are not compatible with the program type codes used in the SIS transport, which could confuse the radio user. However, Genre could possibly be useful in supporting other receiver functions such as scanning or in prompting a receiver to display a genre-based image on the screen.

It is highly recommended to either omit this field or else just broadcast the genre code without additional supporting text.

7.2 Comment Frames

The comment filed is composed of three individual fields: language, short description, and the content field. This may be too complex for some radio displays to have both a short description and a content text field. If a Comment frame is broadcast, the language and short description fields are required to be populated with information. The content field may be a null string (single 0x00 byte) if desired.

7.3 Commercial Frames

Commercial frames are fairly complex and are composed of several individual fields. If a Commercial frame is broadcast, the following rules shall be applied:

- Price string may be a null string (single 0x00 byte). If it is not null, the 3-character currency code is required. After the currency code, the only allowable characters are 0 to 9 and a decimal point. However, a decimal point is not required.
- Valid Until is required. It must be an 8-character date string in the format YYYYMMDD. If a valid date is not desired to be sent, for example, if a date is not applicable to the other content, a value of "00000000" may be used.
- Contact URL may be a null string (single 0x00 byte)
- Received as byte is required: can be any value
- Name of seller may be a null string
- Description may be a null string
- At least one of URL, Seller, and Description shall contain at least one displayable character. It is acceptable for a maximum of two of those fields to be null strings.

7.4 UFID Frames

If a UFID frame is broadcast, it is required to contain information in the Owner Identifier field. The Identifier data field may be empty if it is not needed.

Appendix A

Reference | ID3 web site: http://id3.org/id3v2.3.00

Appendix A provides an annotated version of the ID3 V2.3.0 Standard.

Appendix A shows the portions that are supported by the HD Radio System, not supported by the HD Radio System, and omitted.

ID3 Standard Reference for HD Radio Program Service Data

Informal standard M. Nilsson

Document: id3v2.3.0.txt 3rd February 1999

ID3 tag version 2.3.0

Status of this document

This document is an informal standard and replaces the ID3v2.2.0 standard [ID3v2]. The informal standard is released so that implementers could have a set standard before a formal standard is set. The formal standard will use another version or revision number if not identical to what is described in this document. The contents in this document may change for clarifications but never for added or altered functionality.

Distribution of this document is unlimited.

Abstract

This document describes the ID3v2.3.0, which is a more developed version of the ID3v2 informal standard [ID3v2] (version 2.2.0), evolved from the ID3 tagging system. The ID3v2 offers a flexible way of storing information about an audio file within itself to determine its origin and contents. The information may be technical information, such as equalization curves, as well as related meta information, such as title, performer, copyright etc.

7. References
8. Genre List from ID3v1
9. **omitted**

2. Conventions in this document

In the examples, text within "" is a text string exactly as it appears in a file. Numbers preceded with $ are hexadecimal and numbers preceded with % are binary. $xx is used to indicate a byte with unknown content. %x is used to indicate a bit with unknown content. The most significant bit (MSB) of a byte is called 'bit 7' and the least significant bit (LSB) is called 'bit 0'.

A tag is the whole tag described in this document. A frame is a block of information in the tag. The tag consists of a header, frames and optional padding. A field is a piece of information; one value, a string etc. A numeric string is a string that consists of the characters 0-9 only.

3. ID3v2 overview

The two biggest design goals were to be able to implement ID3v2 without disturbing old software too much and that ID3v2 should be as flexible and expandable as possible.

The first criterion is met by the simple fact that the MPEG [MPEG] decoding software uses a sync signal, embedded in the audio stream, to 'lock on to' the audio. Since the ID3v2 tag doesn't contain a valid sync signal, no software will attempt to play the tag. If, for any reason, coincidence make a sync signal appear within the tag it will be taken care of by the 'unsynchronisation scheme' described in section 5.

The second criterion has made a more noticeable impact on the design of the ID3v2 tag. It is constructed as a container for several information blocks, called frames, whose format need not be known to the software that encounters them. At the start of every frame there is an identifier that explains the frames' format and content, and a size descriptor that allows software to skip unknown frames.

If a total revision of the ID3v2 tag should be needed, there is a version number and a size descriptor in the ID3v2 header.

The ID3 tag described in this document is mainly targeted at files encoded with MPEG-1/2 layer I, MPEG-1/2 layer II, MPEG-1/2 layer III and MPEG-2.5, but may work with other types of encoded audio.

The bit order in ID3v2 is most significant bit first (MSB). The byte order in multi-byte numbers is most significant byte first (e.g. $12345678 would be encoded $12 34 56 78).

It is permitted to include padding after all the final frame (at the end of the ID3 tag), making the size of all the frames together smaller than the size given in the head of the tag. A possible purpose of this padding is to allow for adding a few additional frames or enlarge existing frames within the tag without having to rewrite the entire file. The value of the padding bytes must be $00.

3.1. ID3v2 header

The ID3v2 tag header, which should be the first information in the file, is 10 bytes as follows:

ID3v2/file identifier	"ID3"
ID3v2 version	$03 00
ID3v2 flags	%abc00000
ID3v2 size	4 * %0xxxxxxx

The first three bytes of the tag are always "ID3" to indicate that this is an ID3v2 tag, directly followed by the two version bytes. The first byte of ID3v2 version is it's major version, while the second byte is its revision number. In this case this is ID3v2.3.0. All revisions are backwards compatible while major versions are not. If software with ID3v2.2.0 and below support should encounter version three or higher it should simply ignore the whole tag. Version and revision will never be $FF.

The version is followed by one the ID3v2 flags field, of which currently only three flags are used.

a - Unsynchronisation **Not supported. Must be 0.**

Bit 7 in the 'ID3v2 flags' indicates whether or not unsynchronisation is used (see section 5 for details); a set bit indicates usage.

b - Extended header

The second bit (bit 6) indicates whether or not the header is followed by an extended header. The extended header is described in section 3.2.

c - Experimental indicator

The third bit (bit 5) should be used as an 'experimental indicator'. This flag should always be set when the tag is in an experimental stage.

All the other flags should be cleared. If one of these undefined flags are set that might mean that the tag is not readable for a parser that does not know the flags function.

The ID3v2 tag size is encoded with four bytes where the most significant bit (bit 7) is set to zero in every byte, making a total of 28 bits. The zeroed bits are ignored, so a 257 bytes long tag is represented as $00 00 02 01.

The ID3v2 tag size is the size of the complete tag after unsynchronisation, including padding, excluding the header but not excluding the extended header (total tag size - 10). Only 28 bits (representing up to 256MB) are used in the size description to avoid the introduction of 'false sync signals'.

An ID3v2 tag can be detected with the following pattern:

$49 44 33 yy yy xx zz zz zz zz

Where yy is less than $FF, xx is the 'flags' byte and zz is less than $80.

3.2. ID3v2 extended header

The extended header contains information that is not vital to the correct parsing of the tag information, hence the extended header is optional.

Extended header size $xx xx xx xx

Extended Flags $xx xx

Size of padding $xx xx xx xx **Not supported. Must be $00 00 00 00**

Where the 'Extended header size', currently 6 or 10 bytes, excludes itself. The 'Size of padding' is simply the total tag size excluding the frames and the headers, in other words the padding. The extended header is considered separate from the header proper, and as such is subject to unsynchronisation.

The extended flags are a secondary flag set which describes further attributes of the tag. These attributes are currently defined as follows

%x0000000 00000000

x - CRC data present **Not Supported**

If this flag is set four bytes of CRC-32 data is appended to the extended header. The CRC should be calculated before unsynchronisation on the data between the extended header and the padding, i.e. the frames and only the frames.

Total frame CRC $xx xx xx xx

3.3. ID3v2 frame overview

As the tag consists of a tag header and a tag body with one or more frames, all the frames consists of a frame header followed by one or more fields containing the actual information. The layout of the frame header:

Frame ID $xx xx xx xx (four characters)

Size $xx xx xx xx

Flags $xx xx

The frame ID made out of the characters capital A-Z and 0-9. Identifiers beginning with "X", "Y" and "Z" are for experimental use and free for everyone to use, without the need to set the experimental bit in the tag header. Have in mind that someone else might have used the same identifier as you. All other identifiers are either used or reserved for future use.

The frame ID is followed by a size descriptor, making a total header size of ten bytes in every frame. The size is calculated as frame size excluding frame header (frame size - 10).

In the frame header the size descriptor is followed by two flags bytes. These flags are described in section 3.3.1.

There is no fixed order of the frames' appearance in the tag, although it is desired that the frames are arranged in order of significance concerning the recognition of the file. An example of such order: UFID, TIT2, MCDI, TRCK ...

A tag must contain at least one frame. A frame must be at least 1 byte big, excluding the header.

If nothing else is said a string is represented as ISO-8859-1 [ISO-8859-1] characters in the range $20 - $FF. Such strings are represented as <text string>, or <full text string> if newlines are allowed, in the frame descriptions. All Unicode strings [UNICODE] use 16-bit Unicode 2.0 (ISO/IEC 10646-1:1993, UCS-2). Unicode strings must begin with the Unicode BOM ($FF FE or $FE FF) to identify the byte order.

All numeric strings and URLs [URL] are always encoded as ISO-8859-1. Terminated strings are terminated with $00 if encoded with ISO-8859-1 and $00 00 if encoded as Unicode. If nothing else is said newline character is forbidden. In ISO-8859-1 a new line is represented, when allowed, with $0A only. Frames that allow different types of text encoding have a text encoding description byte directly after the frame size. If ISO-8859-1 is used this byte should be $00, if Unicode is used it should be $01. Strings dependent on encoding is represented as <text string according to encoding>, or <full text string according to encoding> if newlines are allowed. Any empty Unicode strings which are NULL-terminated may have the Unicode BOM followed by a Unicode NULL ($FF FE 00 00 or $FE FF 00 00).

The three byte language field is used to describe the language of the frame's content, according to ISO-639-2 [ISO-639-2].

All URLs [URL] may be relative, e.g. "picture.png", "../doc.txt".

If a frame is longer than it should be, e.g. having more fields than specified in this document, that indicates that additions to the frame have been made in a later version of the ID3v2 standard. This is reflected by the revision number in the header of the tag.

3.3.1. Frame header flags

In the frame header the size descriptor is followed by two flags bytes. All unused flags must be cleared. The first byte is for 'status messages' and the second byte is for encoding purposes. If an unknown flag is set in the first byte the frame may not be changed without the bit cleared. If an unknown flag is set in the second byte it is likely to not be readable. The flags field is defined as follows.

%abc00000 %ijk00000	**Default value = %00000000 %00000000. However, these bits shall be considered reserved for future applications.**

a - Tag alter preservation **Not supported**

This flag tells the software what to do with this frame if it is unknown and the tag is altered in any way. This applies to all kinds of alterations, including adding more padding and reordering the frames.

0 Frame should be preserved.

1 Frame should be discarded.

b - File alter preservation **Not supported**

This flag tells the software what to do with this frame if it is unknown and the file, excluding the tag, is altered. This does not apply when the audio is completely replaced with other audio data.

0 Frame should be preserved.

1 Frame should be discarded.

c - Read only Reserved **Not supported**

This bit is reserved. It may be set or cleared by external subsystems and should therefore be ignored.

i - Compression **Not supported**

This flag indicates whether or not the frame is compressed.

0 Frame is not compressed.

1 Frame is compressed using zlib [zlib] with 4 bytes for 'decompressed size' appended to the frame header.

j - Encryption **Not supported**

This flag indicates whether or not the frame is encrypted. If set one byte indicating with which method it was encrypted will be appended to the frame header. See section 4.26 for more information about encryption method registration.

0 Frame is not encrypted.

1 Frame is encrypted.

k - Grouping identity **Not supported**

This flag indicates whether or not this frame belongs in a group with other frames. If set a group identifier byte is added to the frame header. Every frame with the same group identifier belongs to the same group.

0 Frame does not contain group information

1 Frame contains group information

Some flags indicates that the frame header is extended with additional information. This information will be added to the frame header in the same order as the flags indicating the additions. I.e. the four bytes of decompressed size will precede the encryption method byte. These additions to the frame header, while not included in the frame header size but are included in the 'frame size' field, are not subject to encryption or compression.

3.3.2. Default flags **Omitted**

The default settings for the frames described in this document can be divided into the following classes. The flags may be set differently if found more suitable by the software.

1. Discarded if tag is altered, discarded if file is altered.

   None.

2. Discarded if tag is altered, preserved if file is altered.

   None.

3. Preserved if tag is altered, discarded if file is altered.

   AENC, ETCO, EQUA, MLLT, POSS, SYLT, SYTC, RVAD, TENC, TLEN, TSIZ

4. Preserved if tag is altered, preserved if file is altered.

   The rest of the frames.

4. This section has been omitted.

4.1. Unique file identifier

This frame's purpose is to be able to identify the audio file in a database that may contain more information relevant to the content. Since standardization of such a database is beyond this document, all frames begin with a null-terminated string with a URL [URL] containing an email address, or a link to a location where an email address can be found, that belongs to the organization responsible for this specific database implementation. Questions regarding the database should be sent to the indicated email address. The URL should not be used for the actual database queries. The string "http://www.id3.org/dummy/ufid.html" should be used for tests. Software that isn't told otherwise may safely remove such frames. The 'Owner identifier' must be non-empty (more than just a termination). The 'Owner identifier' is then followed by the actual identifier, which may be up to 64 bytes. There may be more than one "UFID" frame in a tag, but only one with the same 'Owner identifier'.

<Header for 'Unique file identifier', ID: "UFID">

Owner identifier <text string> $00

Identifier <up to 64 bytes binary data>

4.2. Text information frames

The text information frames are the most important frames, containing information like artist, album and more. There may only be one text information frame of its kind in an tag. If the text string is followed by a termination ($00 (00)) all the following information should be ignored and not be displayed. All text frame identifiers begin with "T". Only text frame identifiers begin with "T", with the exception of the "TXXX" frame. All the text information frames have the following format:

<Header for 'Text information frame', ID: "T000" - "TZZZ",

excluding "TXXX" described in 4.2.2.>

Text encoding $xx

Information <text string according to encoding>

4.2.1. Text information frames - details

This section only includes frames supported in MPS.

TALB

The 'Album/Movie/Show title' frame is intended for the title of the recording(/source of sound) which the audio in the file is taken from.

TCON

The 'Content type', which previously was stored as a one byte numeric value only, is now a numeric string. You may use one or several of the types as ID3v1.1 did or, since the category list would be impossible to maintain with accurate and up to date categories, define your own.

References to the ID3v1 genres can be made by, as first byte, enter "(" followed by a number from the genres list (Section 8.1, Appendix A.) and ended with a ")" character. This is optionally followed by a refinement, e.g. "(21)" or "(4) Eurodisco". Several references can be made in the same frame, e.g. "(51)(39)". If the refinement should begin with a "(" character it should be replaced with "((", e.g. "((I can figure out any genre)" or "(55) ((I think...)". The following new content types are defined in ID3v2 and are implemented in the same way as the numeric content types, e.g. "(RX)".

RX Remix

CR Cover

TIT2

The 'Title/Song name/Content description' frame is the actual name of the piece (e.g. "Adagio", "Hurricane Donna").

TPE1

The 'Lead artist(s)/Lead performer(s)/Soloist(s)/Performing group' is

used for the main artist(s). They are separated with the "/"

character.

4.2.2. This section has been omitted.

4.3 thru 4.10 omitted

4.11. Comments

This frame is intended for any kind of full text information that does not fit in any other frame. It consists of a frame header followed by encoding, language and content descriptors and is ended with the actual comment as a text string. Newline characters are allowed in the comment text string. There may be more than one comment frame in each tag, but only one with the same language and content descriptor.

<Header for 'Comment', ID: "COMM">

Text encoding $xx

Language $xx xx xx

Short content descrip. <text string according to encoding> $00 (00)

The actual text <full text string according to encoding>

4.12 thru 4.24 omitted

4.25. Commercial frame

This frame enables several competing offers in the same tag by bundling all needed information. That makes this frame rather complex but it's an easier solution than if one tries to achieve the same result with several frames. The frame begins, after the frame ID, size and encoding fields, with a price string field. A price is constructed by one three character currency code, encoded according to ISO 4217 [ISO-4217] alphabetic currency code, followed by a numerical value where "." is used as decimal separator. In the price string several prices may be concatenated, separated by a "/" character, but there may only be one currency of each type.

The price string is followed by an 8 character date string in the format YYYYMMDD, describing for how long the price is valid. After that is a contact URL, with which the user can contact the seller, followed by a one byte 'received as' field. It describes how the audio is delivered when bought according to the following list:

$00 Other

$01 Standard CD album with other songs

$02 Compressed audio on CD

$03 File over the Internet

$04 Stream over the Internet

$05 As note sheets

$06 As note sheets in a book with other sheets

$07 Music on other media

$08 Non-musical merchandise

Next follows a terminated string with the name of the seller followed by a terminated string with a short description of the product. The last thing is the ability to include a company logotype. The first of them is the 'Picture MIME type' field containing information about which picture format is used. In the event that the MIME media type name is omitted, "image/" will be implied. Currently only "image/png" and "image/jpeg" are allowed. This format string is followed by the binary picture data. This two last fields may be omitted if no picture is to attach.

<Header for 'Commercial frame', ID: "COMR">

Text encoding $xx

Price string <text string> $00

Valid until <text string>

Contact URL <text string> $00

Received as $xx

Name of seller <text string according to encoding> $00 (00)

Description <text string according to encoding> $00 (00)

Picture MIME type <string> $00	**Not Supported**
Seller logo <binary data>	**Not Supported**

4.26. Encryption method registration

To identify with which method a frame has been encrypted the encryption method must be registered in the tag with this frame. The 'Owner identifier' is a null-terminated string with a URL [URL] containing an email address, or a link to a location where an email address can be found, that belongs to the organization responsible for this specific encryption method. Questions regarding the encryption method should be sent to the indicated email address. The 'Method symbol' contains a value that is associated with this method throughout the whole tag. Values below $80 are reserved. The 'Method symbol' may optionally be followed by encryption specific data. There may be several "ENCR" frames in a tag but only one containing the same symbol and only one containing the same owner identifier. The method must be used somewhere in the tag. See section 3.3.1, flag j for more information.

<Header for 'Encryption method registration', ID: "ENCR">

Owner identifier <text string> $00

Method symbol $xx

Encryption data <binary data>

4.27 thru 4.28 omitted

5. The 'unsynchronisation scheme' **Omitted**

The only purpose of the 'unsynchronisation scheme' is to make the ID3v2 tag as compatible as possible with existing software. There is no use in 'unsynchronising' tags if the file is only to be processed by new software. Unsynchronisation may only be made with MPEG 2 layer I, II and III and MPEG 2.5 files.

Whenever a false synchronization is found within the tag, one zeroed byte is inserted after the first false synchronization byte. The format of a correct sync that should be altered by ID3 encoders is as follows:

%11111111 111xxxxx

And should be replaced with:

%11111111 00000000 111xxxxx

This has the side effect that all $FF 00 combinations have to be altered, so they won't be affected by the decoding process. Therefore all the $FF 00 combinations have to be replaced with the $FF 00 00 combination during the unsynchronisation.

To indicate usage of the unsynchronisation, the first bit in 'ID3 flags' should be set. This bit should only be set if the tag contains a, now corrected, false synchronization. The bit should only be clear if the tag does not contain any false synchronizations.

Do bear in mind, that if a compression scheme is used by the encoder, the unsynchronisation scheme should be applied *afterwards*. When decoding a compressed, 'unsynchronized' file, the unsynchronisation scheme' should be parsed first, decompression afterwards.

If the last byte in the tag is $FF, and there is a need to eliminate false synchronizations in the tag, at least one byte of padding should be added.

6. Copyright

7. References

[CDDB] Compact Disc Data Base
<url:http://www.cddb.com>

[ID3v2] Martin Nilsson, "ID3v2 informal standard".
<url:http://www.id3.org/id3v2-00.txt>

[ISO-639-2] ISO/FDIS 639-2.
Codes for the representation of names of languages, Part 2: Alpha-3 code. Technical committee / subcommittee: TC 37 / SC 2

[ISO-4217] ISO 4217:1995.
Codes for the representation of currencies and funds. Technical committee / subcommittee: TC 68

[ISO-8859-1] ISO/IEC DIS 8859-1.
8-bit single-byte coded graphic character sets, Part 1: Latin alphabet No. 1. Technical committee / subcommittee: JTC 1 / SC 2

[ISRC] ISO 3901:1986
International Standard Recording Code (ISRC). Technical committee / subcommittee: TC 46 / SC 9

[JFIF] JPEG File Interchange Format, version 1.02
<url:http://www.w3.org/Graphics/JPEG/jfif.txt>

[MIME] Freed, N. and N. Borenstein, "Multipurpose Internet Mail Extensions (MIME) Part One: Format of Internet Message Bodies", RFC 2045, November 1996.
<url:ftp://ftp.isi.edu/in-notes/rfc2045.txt>

[MPEG] ISO/IEC 11172-3:1993. Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbit/s, Part 3: Audio. Technical committee / subcommittee: JTC 1 / SC 29
and
ISO/IEC 13818-3:1995 Generic coding of moving pictures and associated audio information, Part 3: Audio. Technical committee / subcommittee: JTC 1 / SC 29
and
ISO/IEC DIS 13818-3 Generic coding of moving pictures and associated audio information, Part 3: Audio (Revision of ISO/IEC 13818-3:1995)

[PNG] Portable Network Graphics, version 1.0
<url:http://www.w3.org/TR/REC-png-multi.html>

[UNICODE] ISO/IEC 10646-1:1993. Universal Multiple-Octet Coded Character Set (UCS), Part 1: Architecture and Basic Multilingual Plane. Technical committee / subcommittee: JTC 1 / SC 2
url:http://www.unicode.org

[URL] T. Berners-Lee, L. Masinter & M. McCahill, "Uniform Resource Locators (URL).", RFC 1738, December 1994.
<url:ftp://ftp.isi.edu/in-notes/rfc1738.txt>

[ZLIB] P. Deutsch, Aladdin Enterprises & J-L. Gailly, "ZLIB Compressed Data Format Specification version 3.3", RFC 1950, May 1996.
<url:ftp://ftp.isi.edu/in-notes/rfc1950.txt>

8. Appendix

8.1 Appendix A – Genre List from ID3v1

The following genres is defined in ID3v1		The following genres are Winamp extensions
0.Blues	46.Instrumental Pop	80.Folk
1.Classic Rock	47.Instrumental Rock	81.Folk-Rock
2.Country	48.Ethnic	82.National Folk
3.Dance	49.Gothic	83.Swing
4.Disco	50.Darkwave	84.Fast Fusion
5.Funk	51.Techno-Industrial	85.Bebob
6.Grunge	52.Electronic	86.Latin
7.Hip-Hop	53.Pop-Folk	87.Revival
8.Jazz	54.Eurodance	88.Celtic
9.Metal	55.Dream	89.Bluegrass
10.New Age	56.Southern Rock	90.Avantgarde
11.Oldies	57.Comedy	91.Gothic Rock
12.Other	58.Cult	92.Progressive Rock
13.Pop	59.Gangsta	93.Psychedelic Rock
14.R&B	60.Top 40	94.Symphonic Rock
15.Rap	61.Christian Rap	95.Slow Rock
16.Reggae	62.Pop/Funk	96.Big Band
17.Rock	63.Jungle	97.Chorus
18.Techno	64.Native American	98.Easy Listening
19.Industrial	65.Cabaret	99.Acoustic
20.Alternative	66.New Wave	100.Humour
21.Ska	67.Psychadelic	101.Speech
22.Death Metal	68.Rave	102.Chanson
23.Pranks	69.Showtunes	103.Opera
24.Soundtrack	70.Trailer	104.Chamber Music
25.Euro-Techno	71.Lo-Fi	105.Sonata
26.Ambient	72.Tribal	106.Symphony
27.Trip-Hop	73.Acid Punk	107.Booty Bass
28.Vocal	74.Acid Jazz	108.Primus
29.Jazz+Funk	75.Polka	109.Porn Groove
30.Fusion	76.Retro	110.Satire
31.Trance	77.Musical	111.Slow Jam
32.Classical	78.Rock & Roll	112.Club
33.Instrumental	79.Hard Rock	113.Tango
34.Acid		114.Samba
35.House		115.Folklore
36.Game		116.Ballad
37.Sound Clip		117.Power Ballad
38.Gospel		118.Rhythmic Soul
39.Noise		119.Freestyle
40.AlternRock		120.Duet
41.Bass		121.Punk Rock
42.Soul		122.Drum Solo
43.Punk		123.Acapella
44.Space		124.Euro-House
45.Meditative		125.Dance Hall

9. This section has been omitted

HD Radio™ AM Transmission

System Specifications

Revision H

November 2022

SY_SSS_1082s

TRADEMARKS

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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 details specifications of the iBiquity Digital Corporation HD Radio AM IBOC system. Included in this document are specifications that ensure reliable reception of the digital audio and data, provide precise digital-analog synchronization, define subcarrier power levels, and minimize harmful spectral emissions.

2 Reference Documents

STATEMENT
Each referenced document that is mentioned in this document shall be listed in the following iBiquity document:

- Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard Document Number: SY_REF_2690s

3 Abbreviations, Acronyms, and Conventions

3.1 Abbreviations and Acronyms

AM	Amplitude Modulation
BPSK	Binary Phase Shift Keying
FCC	Federal Communications Commission
FM	Frequency Modulation
GPS	Global Positioning System
IBOC	In-Band On-Channel
kbit/s	kilobits per second (thousand bits per second)
L1	Layer 1
L2	Layer 2
MER	Modulation Error Ratio
MF	Medium Frequency
MA1	Primary AM Hybrid Service Mode
MA3	Primary AM All Digital Service Mode
N/A	Not Applicable
NRSC	National Radio Systems Committee
OFDM	Orthogonal Frequency Division Multiplexing
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RF	Radio Frequency
SSB	Single Side Band
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.
- Bit 0 of a byte or word is considered the least significant bit.
- In representations of binary numbers, the least significant bit is on the right.
- 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.

3.3 Arithmetic Operators

The arithmetic operators used throughout this document are defined below:

Category	Definition	Examples
| x |	Indicates the absolute value of x	| -5 | = 5 | 3 - 4 | = 1

4 AM Transmission Specifications

4.1 Introduction

This document presents the key transmission specifications for the AM HD Radio system.

4.2 Carrier Frequency and Channel Spacing

The HD Radio system operates in-band and on-channel, within the existing allocations and channel spacing as authorized by the FCC in accordance with [12]. The Hybrid and All Digital HD Radio waveforms are centered on the assigned AM band channel frequency.

4.3 Synchronization Tolerances

The system supports two levels of synchronization for broadcasters:

Level I: Network Synchronized (assumed using Global Positioning System (GPS) locked transmission facilities)

Level II: Non-networked Synchronized (non-GPS-locked transmission facilities)

It is recommended that transmission facilities operate as Level I facilities in order to support numerous advanced system features.

4.3.1 Analog Diversity Delay

The absolute accuracy of the analog diversity delay as defined in [2] in the transmission signal shall be within ±68 microseconds (μs) for both Synchronization Level I and Level II transmission facilities. This is equivalent to ±3 audio samples at a sampling rate of 44.1 kHz.

4.3.1.1 Analog Diversity Delay Correction

Correction of the analog diversity delay may be accomplished either in one large step or by smoothly ramping the delay. Under no circumstances shall a delay ramp rate exceed 50 samples per second. When not implementing a correction, the maximum rate of change of the analog diversity delay shall not exceed 3 audio samples (68 μs) per 10 seconds. Refer to Reference [38] for additional guidance on recommended best practices.

4.3.2 Time and Frequency Accuracy and Stability

The total modulation symbol-clock frequency absolute error of an HD Radio broadcast system shall meet the following requirements:

±0.01 ppm maximum for Synchronization Level I facilities

±1.0 ppm maximum for Synchronization Level II facilities

The total carrier frequency absolute error shall meet the following requirements:

The total (analog and digital) carrier frequency absolute error of a Synchronization Level I broadcast system as observed at the RF output shall be ±0.02 Hz maximum.

The total (analog and digital) carrier frequency absolute error of a Synchronization Level II broadcast system as observed at the RF output shall be ±2.0 Hz maximum.

4.3.3 L1 Frame Timing Phase

For Level I transmission facilities, all transmissions shall phase lock their L1 frame timing (and the timing of all OFDM symbols) to absolute GPS time within ±1 μs.

If the above specification in Synchronization Level I transmission facility is violated, due to a GPS outage or other occurrence, it shall be classified as a Synchronization Level II transmission facility until the above specification is again met.

4.4 AM Analog Host Performance (Hybrid Transmissions)

Hybrid service mode MA1 may be broadcast in one of several configurations. Normally, all of the primary, secondary, and tertiary subcarriers are enabled. In this case, some of the digital sidebands are superimposed in the same spectrum as the analog host signal. There are two different configurations available for this case. In the 5 kHz analog audio bandwidth configuration, the analog host shares the same spectrum as the tertiary subcarriers. In the 8 kHz analog audio bandwidth configuration, the analog host also shares its spectrum with a portion of the secondary subcarriers.

In addition, there is a reduced digital bandwidth configuration where the secondary and tertiary subcarriers are shut off. This configuration is selected by setting the RDB control signal to 1 as explained in [2]. In the reduced digital bandwidth configuration, it is possible to extend the analog audio bandwidth up to 9.4 kHz since there is no potential interference to the secondary or tertiary subcarriers. However, as explained in [30], there are various reasons why extending the analog audio bandwidth beyond 5 kHz is not recommended.

For each of the three configurations just described, the following information applies:

> The analog signal shall meet the FCC emissions mask specifications contained in 47 CFR §73.44.
>
> It is recommended that the host analog audio source be filtered according to the guidelines in [13]. In addition, the following RF performance specifications shall be met:
>
>> For Hybrid transmissions configured for the 5 kHz analog audio bandwidth configuration, the power spectral density of the modulated AM carrier measured with the HD Radio digital component disabled, at frequencies removed from the carrier frequency by more than 5 kHz and up to 20 kHz shall not exceed -65 dBc/300 Hz.
>>
>> For Hybrid transmissions configured for the 8 kHz analog audio bandwidth configuration, the power spectral density of the modulated AM carrier measured with the HD Radio digital component disabled, at frequencies removed from the carrier frequency by more than 8 kHz and up to 20 kHz shall not exceed -65 dBc/300 Hz.
>>
>> For Hybrid transmissions configured for the reduced digital bandwidth configuration (RDB=1), the power spectral density of the modulated AM carrier measured with the HD Radio digital component disabled, at frequencies removed from the carrier frequency by more than 9.4 kHz and up to 20 kHz shall not exceed -65 dBc/300 Hz.
>>
>> Zero dBc is defined as the total power of the unmodulated AM carrier.

4.5 AM Spectral Emissions Limits

The spectral emissions limits for Hybrid transmissions are given in Subsections 4.5.1, 4.5.2, and 4.5.3.

The spectral emissions limits for All Digital transmissions are given in Subsection 4.5.4.

4.5.1 Spectral Emissions Limits for Hybrid Transmissions with 5 kHz Analog Bandwidth Configuration

For Hybrid transmissions, measurements of the combined analog and digital signals shall be made by averaging the power spectral density of the signal in a 300-Hz bandwidth over a minimum time span of 30 seconds and a minimum of 100 sweeps. The measurement point and the test configuration shall be as described in Reference [26].

Zero dBc is defined as the total power of the unmodulated analog AM carrier.

Under normal operation with analog modulation present, the secondary and tertiary subcarriers enabled (RDB=0), and the analog audio bandwidth limited to 5 kHz, the following requirements shall be met at all times. These requirements are applicable for all states of the High Power PIDS (HPP) and Power Level (PL) controls described in [2].

Noise and spuriously generated signals from all sources, including phase noise and intermodulation products, shall conform to the limits as described in the following paragraph and shown in Figure 4-1 and Table 4-1*. These limits are applicable for all permissible power levels of the upper and lower sidebands, as defined in Subsection 4.6.

The measured power spectral density at frequencies greater than 5 kHz, up to and including 9.4 kHz, from the carrier frequency shall not exceed -34.3 dBc/300 Hz.

The measured power spectral density at frequencies greater than 9.4 kHz, up to and including 15 kHz, from the carrier frequency shall not exceed -26.8 dBc/300 Hz.

The measured power spectral density at frequencies greater than 15 kHz, up to and including 15.2 kHz, from the carrier frequency shall not exceed -28 dBc/300 Hz.

The measured power spectral density of the Hybrid signal at frequencies removed from the carrier frequency by more than 15.2 kHz, up to and including 15.8 kHz shall not exceed -39 - (|offset frequency in kHz| - 15.2) • 43.3 dBc/300 Hz.

The measured power spectral density of the Hybrid signal at frequencies removed from the carrier frequency by more than 15.8 kHz, up to and including 25 kHz shall not exceed -65 dBc/300 Hz.

The measured power spectral density of the Hybrid signal at frequencies removed from the carrier frequency by more than 25 kHz, up to and including 30.5 kHz shall not exceed -65 - (|offset frequency in kHz| - 25) • 1.273 dBc/300 Hz.

The measured power spectral density of the Hybrid signal at frequencies removed from the carrier frequency by more than 30.5 kHz, up to and including 75 kHz shall not exceed -72 - (|offset frequency in kHz| - 30.5) • 0.292 dBc/300 Hz.

The measured power spectral density of the Hybrid signal at frequencies removed from the carrier frequency by more than 75 kHz, shall not exceed -85 dBc/300 Hz.

If discrete components exceed the limits established in Table 4-1 and in Figure 4-1, the following conditions shall be met when averaging the power spectral density of the signal in each 300-Hz bandwidth over a minimum time span of 30 seconds and a minimum of 100 sweeps.

1. No more than two discrete components within 75 kHz of the carrier frequency shall exceed the spectral emission limits by more than 10 dB.
2. No more than four discrete components removed from the carrier frequency by more than 75 kHz shall exceed the spectral emission limits by more than 5 dB.

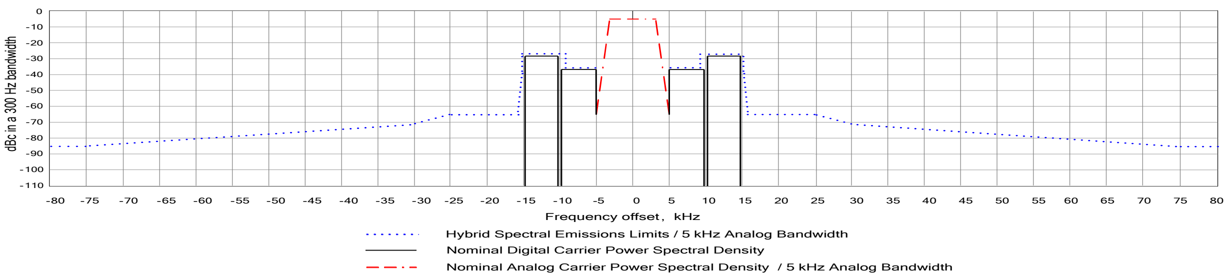

Figure 4-1: HD Radio AM Hybrid Waveform Spectral Emissions Limits for 5 kHz Analog Bandwidth Configuration

Table 4-1: HD Radio AM Hybrid Waveform Spectral Emissions Limits for 5 kHz Analog Bandwidth Configuration*

Frequency Offset Relative to Carrier	**Level Relative to Unmodulated Carrier (dBc per 300 Hz)**
5 to 9.4 kHz offset	-34.3
9.4 to 15 kHz offset	-26.8
15 to 15.2 kHz offset	-28
15.2 to 15.8 kHz offset	-39 - (|frequency offset in kHz| - 15.2) · 43.3
15.8 to 25 kHz offset	-65
25 kHz to 30.5 kHz offset	-65 - (|frequency offset in kHz| - 25) · 1.273
30.5 kHz to 75 kHz offset	-72 - (|frequency offset in kHz| - 30.5) · 0.292
> 75 kHz offset	-85

* The requirements for noise and spurious emission limits defined in this subsection reflect acceptable performance criteria. In certain circumstances, additional measures may be needed to reduce the spectral emissions below the limits given in this subsection in order to reduce mutual interference between broadcast stations.

4.5.2 Spectral Emissions Limits for Hybrid Transmissions with 8 kHz Analog Bandwidth Configuration

For hybrid transmissions, measurements of the combined analog and digital signals shall be made by averaging the power spectral density of the signal in a 300-Hz bandwidth over a minimum time span of 30 seconds and a minimum of 100 sweeps. The measurement point and the test configuration shall be as described in Reference [26].

Zero dBc is defined as the total power of the unmodulated analog AM carrier.

Under normal operation with analog modulation present, the secondary and tertiary subcarriers enabled (RDB=0), and the analog audio bandwidth limited to 8 kHz, the following requirements shall be met at all times. These requirements are applicable for all states of the High Power PIDS (HPP) and Power Level (PL) controls described in [2].

Noise and spuriously generated signals from all sources, including phase noise and intermodulation products, shall conform to the limits as described in the following paragraph and shown in Figure 4-2 and Table 4-2†. These limits are applicable for all permissible power levels of the upper and lower sidebands, as defined in Subsection 4.6.

The measured power spectral density at frequencies greater than 8 kHz, up to and including 9.4 kHz, from the carrier frequency shall not exceed -34.3 dBc/300 Hz.

The measured power spectral density at frequencies greater than 9.4 kHz, up to and including 15 kHz, from the carrier frequency shall not exceed -26.8 dBc/300 Hz.

The measured power spectral density at frequencies greater than 15 kHz, up to and including 15.2 kHz, from the carrier frequency shall not exceed -28 dBc/300 Hz.

The measured power spectral density of the hybrid signal at frequencies removed from the carrier frequency by more than 15.2 kHz, up to and including 15.8 kHz shall not exceed -39 - (|offset frequency in kHz| - 15.2) • 43.3 dBc/300 Hz.

The measured power spectral density of the hybrid signal at frequencies removed from the carrier frequency by more than 15.8 kHz, up to and including 25 kHz shall not exceed -65 dBc/300 Hz.

The measured power spectral density of the hybrid signal at frequencies removed from the carrier frequency by more than 25 kHz, up to and including 30.5 kHz shall not exceed -65 - (|offset frequency in kHz| - 25) • 1.273 dBc/300 Hz.

The measured power spectral density of the hybrid signal at frequencies removed from the carrier frequency by more than 30.5 kHz, up to and including 75 kHz shall not exceed -72 - (|offset frequency in kHz| - 30.5) • 0.292 dBc/300 Hz.

The measured power spectral density of the hybrid signal at frequencies removed from the carrier frequency by more than 75 kHz, shall not exceed -85 dBc/300 Hz.

If discrete components exceed the limits established in Table 4-2 and in Figure 4-2, the following conditions shall be met when averaging the power spectral density of the signal in each 300-Hz bandwidth over a minimum time span of 30 seconds and a minimum of 100 sweeps:

1. No more than two discrete components within 75 kHz of the carrier frequency shall exceed the spectral emission limits by more than 10 dB.
2. No more than four discrete components removed from the carrier frequency by more than 75 kHz shall exceed the spectral emission limits by more than 5 dB.

When a station operates in the Hybrid 8 kHz configuration, an HD Radio receiver will treat the enhanced carriers as complimentary. Complimentary carriers require that both the upper and lower sidebands be recovered for demodulation. Therefore, in the 8 kHz configuration, digital coverage of a Hybrid station may be adversely impacted by adjacent transmission. The severity of the impact will be dependent upon whether the interference is from a first or second adjacent and if it is an Analog, Hybrid or All Digital transmission.

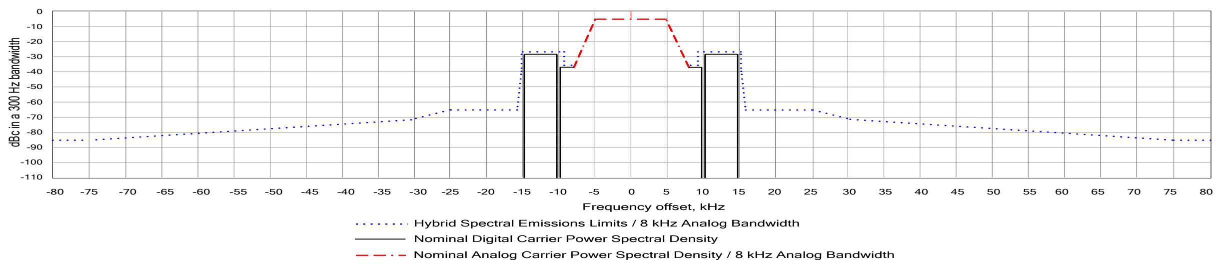

Figure 4-2: HD Radio AM Hybrid Waveform Spectral Emissions Limits for 8 kHz Analog Bandwidth Configuration

Table 4-2: HD Radio AM Hybrid Waveform Spectral Emissions Limits for 8 kHz Analog Bandwidth Configuration†

Frequency Offset Relative to Carrier	**Level Relative to Unmodulated Carrier (dBc per 300 Hz)**
8 to 9.4 kHz offset	-34.3
9.4 to 15 kHz offset	-26.8
15 to 15.2 kHz offset	-28
15.2 to 15.8 kHz offset	-39 - (|frequency offset in kHz| - 15.2) · 43.3
15.8 to 25 kHz offset	-65
25 kHz to 30.5 kHz offset	-65 - (|frequency offset in kHz| - 25) · 1.273
30.5 kHz to 75 kHz offset	-72 - (|frequency offset in kHz| - 30.5) · 0.292
> 75 kHz offset	-85

† The requirements for noise and spurious emission limits defined in this subsection reflect acceptable performance criteria. In certain circumstances, additional measures may be needed to reduce the spectral emissions below the limits given in this subsection in order to reduce mutual interference between broadcast stations.

4.5.3 Spectral Emissions Limits for Hybrid Transmissions with Reduced Digital Bandwidth Configuration

As described in [2], the system provides for a reduced digital bandwidth configuration where the secondary and tertiary subcarriers are disabled. This configuration is selected by setting the control signal RDB to 1. This subsection discusses the spectral emissions limits for such a configuration.

For hybrid transmissions, measurements of the combined analog and digital signals shall be made by averaging the power spectral density of the signal in a 300-Hz bandwidth over a minimum time span of 30 seconds and a minimum of 100 sweeps. The measurement point and the test configuration shall be as described in Reference [26].

Zero dBc is defined as the total power of the unmodulated analog AM carrier.

In the reduced digital bandwidth configuration, under normal operation with analog modulation present and the analog audio bandwidth limited to no more than 9.4 kHz, the following requirements shall be met at all times:

Noise and spuriously generated signals from all sources, including phase noise and intermodulation products, shall conform to the limits as described in the following paragraph and shown in Figure 4-4 and Table 4-4†. These limits are applicable for all permissible power levels of the upper and lower sidebands, as defined in Subsection 4.6.

The measured power spectral density at frequencies greater than 9.4 kHz, up to and including 15 kHz, from the carrier frequency shall not exceed -26.8 dBc/300 Hz.

The measured power spectral density at frequencies greater than 15 kHz, up to and including 15.2 kHz, from the carrier frequency shall not exceed -28 dBc/300 Hz.

The measured power spectral density of the hybrid signal at frequencies removed from the carrier frequency by more than 15.2 kHz, up to and including 15.8 kHz shall not exceed -39 - (|offset frequency in kHz| - 15.2) • 43.3 dBc/300 Hz.

The measured power spectral density of the hybrid signal at frequencies removed from the carrier frequency by more than 15.8 kHz, up to and including 25 kHz shall not exceed -65 dBc/300 Hz.

The measured power spectral density of the hybrid signal at frequencies removed from the carrier frequency by more than 25 kHz, up to and including 30.5 kHz shall not exceed -65 - (|offset frequency in kHz| - 25) • 1.273 dBc/300 Hz.

The measured power spectral density of the hybrid signal at frequencies removed from the carrier frequency by more than 30.5 kHz, up to and including 75 kHz shall not exceed -72 - (|offset frequency in kHz| - 30.5) • 0.292 dBc/300 Hz.

The measured power spectral density of the hybrid signal at frequencies removed from the carrier frequency by more than 75 kHz, shall not exceed -85 dBc/300 Hz.

If discrete components exceed the limits established in Table 4-2 and in Figure 4-2, the following conditions shall be met when averaging the power spectral density of the signal in each 300-Hz bandwidth over a minimum time span of 30 seconds and a minimum of 100 sweeps:

1. No more than two discrete components within 75 kHz of the carrier frequency shall exceed the spectral emission limits by more than 10 dB.
2. No more than four discrete components removed from the carrier frequency by more than 75 kHz shall exceed the spectral emission limits by more than 5 dB.

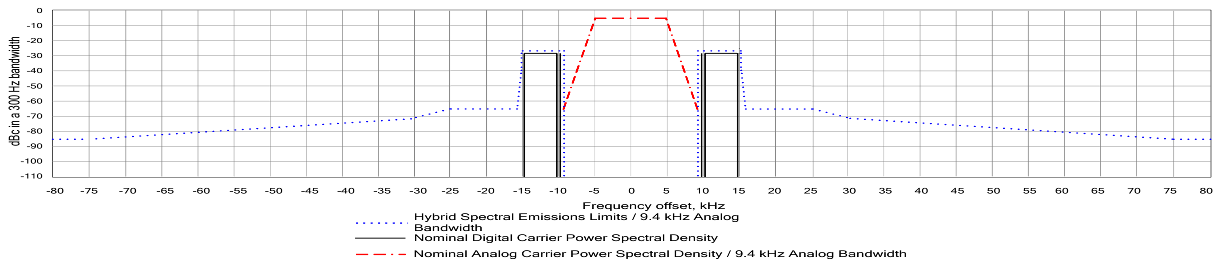

Figure 4-3: HD Radio AM Hybrid Waveform Spectral Emissions Limits for RDB=1 Configuration

Table 4-3: HD Radio AM Hybrid Waveform Spectral Emissions Limits for RDB=1 Configuration†

Frequency Offset Relative to Carrier	Level Relative to Unmodulated Carrier (dBc per 300 Hz)
9.4 to 15 kHz offset	-26.8
15 to 15.2 kHz offset	-28
15.2 to 15.8 kHz offset	-39 - (|frequency offset in kHz| - 15.2) · 43.3
15.8 to 25 kHz offset	-65
25 kHz to 30.5 kHz offset	-65 - (|frequency offset in kHz| - 25) · 1.273
30.5 kHz to 75 kHz offset	-72 - (|frequency offset in kHz| - 30.5) · 0.292
> 75 kHz offset	-85

† The requirements for noise and spurious emission limits defined in this subsection reflect acceptable performance criteria. In certain circumstances, additional measures may be needed to reduce the spectral emissions below the limits given in this subsection in order to reduce mutual interference between broadcast stations.

4.5.4 Spectral Emissions Limits for All Digital Transmissions

For All Digital transmissions, measurements of the All Digital signal shall be made by averaging the power spectral density in a 300-Hz bandwidth over a minimum time span of 30 seconds and a minimum of 100 sweeps. The measurement point and the test configuration shall be as described in Reference [26].

Zero dBc is defined as the allocated power of the unmodulated analog AM carrier and is equal to the reference level used in Subsection 4.5.1.

Under normal operation, the following requirements shall be met at all times. These requirements apply regardless of the waveform configuration; i.e., the spectral emissions limits are applicable for all states of the High Power PIDS (HPP) and Power Level (PL) controls.

Noise and spuriously generated signals from all sources including phase noise and intermodulation products, shall conform to the limits as described in the following paragraph and as shown in Figure 4-4 and Table 4-4‡.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 0.3 kHz, up to and including 5.0 kHz shall not exceed -12.3 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 5.0 kHz, up to and including 5.9 kHz shall not exceed -12.3 - (|frequency offset in kHz| - 5.0) · 16.67 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 5.9 kHz, up to and including 10.0 kHz shall not exceed -27.3 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 10.0 kHz, up to and including 11.2 kHz shall not exceed -27.3 - (|frequency offset in kHz| - 10.0) · 23.08 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 11.2 kHz, up to and including 20.0 kHz shall not exceed -55 - (|frequency offset in kHz| - 11.2) · 1.25 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 20.0 kHz, up to and including 30.0 kHz shall not exceed -66 - (|frequency offset in kHz| - 20.0) · 0.6 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 30.0 kHz, up to and including 60.0 kHz shall not exceed -72 - (|frequency offset in kHz| - 30) · 0.27 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 60 kHz, shall not exceed -80 dBc/300 Hz.

If discrete components exceed the limits established in Table 4-4 and in Figure 4-4, the following conditions shall be met when averaging the power spectral density of the signal in each 300-Hz bandwidth over a minimum time span of 30 seconds and a minimum of 100 sweeps:

1. No more than two discrete components within 75 kHz of the carrier frequency shall exceed the spectral emission limits by more than 10 dB.
2. No more than four discrete components removed from the carrier frequency by more than 75 kHz shall exceed the spectral emission limits by more than 5 dB.

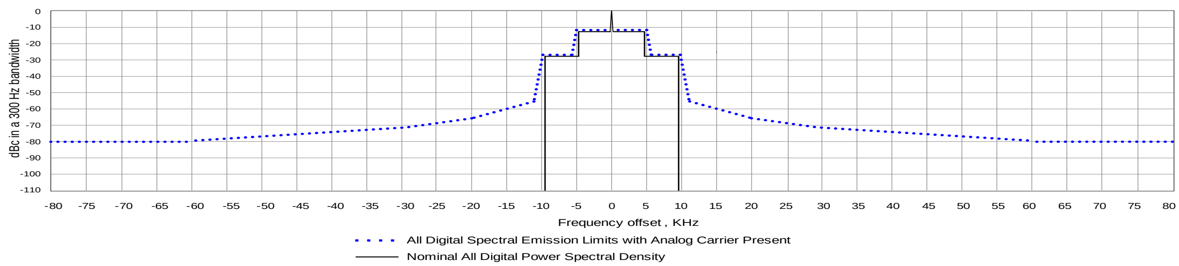

Figure 4-4: HD Radio AM All Digital Waveform Spectral Emissions Limits

Table 4-4: HD Radio AM All Digital Waveform Spectral Emissions Limits‡

Frequency Offset Relative to Carrier	**Level Relative to Unmodulated Carrier (dBc per 300 Hz)**
0.3 kHz to 5.0 kHz offset	-12.3
5.0 kHz to 5.9 kHz offset	-12.3 - (|frequency offset in kHz| - 5.0) · 16.67
5.9 kHz to 10.0 kHz offset	-27.3
10.0 to 11.2 kHz offset	-27.3 - (|frequency offset in kHz| - 10.0) · 23.08
11.2 to 20.0 kHz offset	-55 - (|frequency offset in kHz| - 11.2) · 1.25
20.0 to 30.0 kHz offset	-66 - (|frequency offset in kHz| - 20.0) · 0.6
30.0 to 60.0 kHz offset	-72 - (|frequency offset in kHz| - 30) · 0.27
> 60 kHz offset	-80

‡ The requirements for noise and spurious emission limits defined in this subsection reflect acceptable performance criteria. In certain circumstances, additional measures may be needed to reduce the spectral emissions below the limits given in this subsection in order to reduce mutual interference between broadcast stations.

4.5.5 Spectral Emissions Limits for All Digital Transmissions with Reduced Digital Bandwidth Configuration

As described in [2], the system provides for a reduced digital bandwidth configuration where the secondary and tertiary subcarriers are disabled. This configuration is selected by setting the control signal RDB to 1. This subsection discusses the spectral emissions limits for such a configuration.

For All Digital transmissions in the reduced digital bandwidth configuration, measurements of the All Digital signal shall be made by averaging the power spectral density in a 300-Hz bandwidth over a minimum time span of 30 seconds and a minimum of 100 sweeps. The measurement point and the test configuration shall be as described in Reference [26].

Zero dBc is defined as the allocated power of the unmodulated analog AM carrier and is equal to the reference level used in Subsection 4.5.1.

Under normal operation, the following requirements shall be met at all times. These requirements apply regardless of the waveform configuration; i.e., the spectral emissions limits are applicable for all states of the High Power PIDS (HPP) and Power Level (PL) controls.

Noise and spuriously generated signals from all sources including phase noise and intermodulation products, shall conform to the limits as described in the following paragraph and as shown in Figure 4-5 and Table 4-5‡.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 0.3 kHz, up to and including 5.0 kHz shall not exceed -12.3 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 5.0 kHz, up to and including 7.0 kHz shall not exceed -12.3 - (|frequency offset in kHz| - 5.0) · 17.35 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 7.0 kHz, up to and including 10.4 kHz shall not exceed -47 - (|frequency offset in kHz| - 7.0) · 2.06 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 10.4 kHz, up to and including 20.0 kHz shall not exceed -54 - (|frequency offset in kHz| - 10.4) · 1.25 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 20.0 kHz, up to and including 30.0 kHz shall not exceed -66 - (|frequency offset in kHz| - 20.0) · 0.60 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 30.0 kHz, up to and including 60.0 kHz shall not exceed -72 - (|frequency offset in kHz| - 30.0) · 0.27 dBc/300 Hz.

The measured power spectral density of the All Digital signal at frequencies removed from the carrier frequency by more than 60 kHz, shall not exceed -80 dBc/300 Hz.

If discrete components exceed the limits established in Table 4-5 and in Figure 4-5, the following conditions shall be met when averaging the power spectral density of the signal in each 300-Hz bandwidth over a minimum time span of 30 seconds and a minimum of 100 sweeps:

1. No more than two discrete components within 75 kHz of the carrier frequency shall exceed the spectral emission limits by more than 10 dB.
2. No more than four discrete components removed from the carrier frequency by more than 75 kHz shall exceed the spectral emission limits by more than 5 dB.

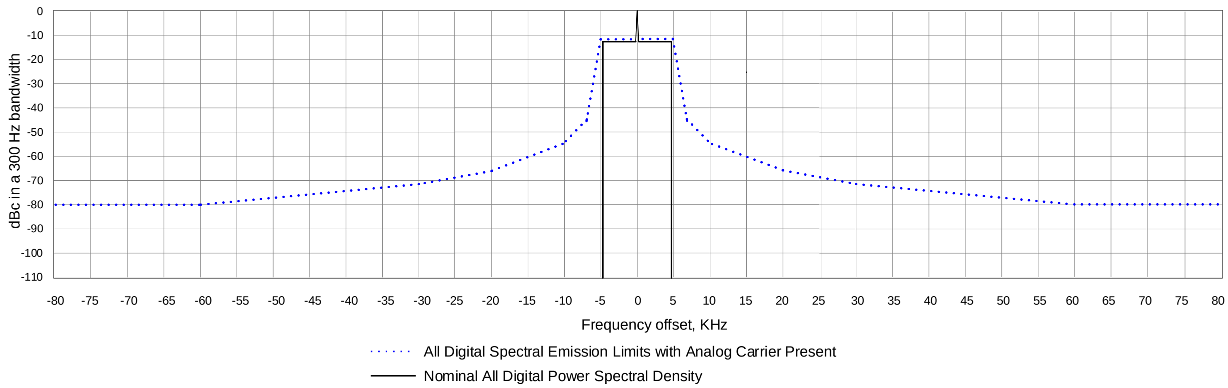

Figure 4-5: HD Radio AM All Digital Waveform Spectral Emissions Limits for RDB=1 Configuration

Table 4-5: HD Radio AM All Digital Waveform Spectral Emissions Limits for RDB=1 Configuration ‡

Frequency Offset Relative to Carrier	**Level Relative to Unmodulated Carrier (dBc per 300 Hz)**
0.3 kHz to 5.0 kHz offset	-12.3
5.0 kHz to 7.0 kHz offset	-12.3 - (|frequency offset in kHz| - 5.0) · 17.35
7.0 to 10.4 kHz offset	-47 - (|frequency offset in kHz| - 7.0) · 2.06
10.4 to 20.0 kHz offset	-54 - (|frequency offset in kHz| - 10.4) · 1.25
20.0 to 30.0 kHz offset	-66 - (|frequency offset in kHz| - 20.0) · 0.60
30.0 to 60.0 kHz offset	-72 - (|frequency offset in kHz| - 30.0) · 0.27
> 60 kHz offset	-80

‡ The requirements for noise and spurious emission limits defined in this subsection reflect acceptable performance criteria. In certain circumstances, additional measures may be needed to reduce the spectral emissions below the limits given in this subsection in order to reduce mutual interference between broadcast stations.

4.6 Digital Sideband Levels

The amplitude scaling of each OFDM subcarrier within each digital sideband is given in Table 4-6 for the Hybrid and All Digital waveforms. The amplitude scale factors are such that the average power in the constellation for that subcarrier meets the average per subcarrier power spectral density shown in dB.

For both the Hybrid and All Digital waveforms, the subcarrier levels are specified relative to the total power of the unmodulated analog AM carrier (assumed equal to 1).

Refer to [2] for a description of how the various scale factors are selected and applied to the various waveforms.

In service mode MA1, the power of one primary sideband may be scaled downward if necessary to reduce potential interference to another broadcast on an adjacent channel. However, all of the other sidebands must maintain the levels shown in Table 4-6.

In service mode MA3, asymmetric sideband operation is not permitted.

Refer to Figure 4-6 through Figure 4-10 for illustrations of how the scale factors are applied for each of the service mode MA1 configurations. In each of these figures, the typical case of symmetric sideband operation is shown.

Optionally, asymmetric sideband operation is permissible for each of the MA1 configurations. Refer to Figure 4-11 for an illustration of asymmetric sideband operation. Such operation is possible in all MA1 configurations. However, only the configuration of RDB=0, PL-0, and HPP=0 is shown. Note that the outer PIDS subcarriers are scaled according to the individual primary sideband levels.

Refer to Figure 4-12 through Figure 4-14 for illustrations of how the scale factors are applied for each of the service mode MA3 configurations.

Table 4-6: OFDM Subcarrier Amplitude Scaling

Waveform	Service Mode	Sidebands	Amplitude Scale Factor Notation	Modulation Type	Nominal Power Spectral Density, dBc/Subcarrier	Nominal Power Spectral Density in a 300 Hz Bandwidth, dBc
Hybrid	MA1	Primary Upper	CH_{PU}	64-QAM	-30 (Note 1, 2)	-27.8
		Primary Lower	CH_{PL}	64-QAM	-30 (Note 1, 2)	-27.8
		Secondary Upper	CH_{S1}	16-QAM	-43	-40.8
			CH_{S2}	16-QAM	-37	-34.8
		Secondary Lower	CH_{S1}	16-QAM	-43	-40.8
			CH_{S2}	16-QAM	-37	-34.8
		Tertiary Upper	$\underline{CH}_{T1}$ [0]	QPSK	-44	-41.8
			$\underline{CH}_{T1}$ [1]	QPSK	-44.5	-42.8
			$\underline{CH}_{T1}$ [2]	QPSK	-45	-42.8
			$\underline{CH}_{T1}$ [3]	QPSK	-45.5	-43.3
			$\underline{CH}_{T1}$ [4]	QPSK	-46	-43.8
			$\underline{CH}_{T1}$ [5]	QPSK	-46.5	-44.3
			$\underline{CH}_{T1}$ [6]	QPSK	-47	-44.8

Waveform	Service Mode	Sidebands	Amplitude Scale Factor Notation	Modulation Type	Nominal Power Spectral Density, dBc/Subcarrier	Nominal Power Spectral Density in a 300 Hz Bandwidth, dBc
			$\underline{CH}_{T1}$ [7]	QPSK	-47.5	-45.3
			$\underline{CH}_{T1}$ [8]	QPSK	-48	-45.8
			$\underline{CH}_{T1}$ [9]	QPSK	-48.5	-46.3
			$\underline{CH}_{T1}$ [10]	QPSK	-49	-46.8
			$\underline{CH}_{T1}$ [11]	QPSK	-49.5	-47.3
			$\underline{CH}_{T1}$ [12:24]	QPSK	-50	-47.8
			$\underline{CH}_{T2}$ [0:24]	QPSK	-44	-41.8
		Tertiary Lower	$\underline{CH}_{T1}$ [0]	QPSK	-44	-41.8
			$\underline{CH}_{T1}$ [1]	QPSK	-44.5	-42.8
			$\underline{CH}_{T1}$ [2]	QPSK	-45	-42.8
			$\underline{CH}_{T1}$ [3]	QPSK	-45.5	-43.3
			$\underline{CH}_{T1}$ [4]	QPSK	-46	-43.8
			$\underline{CH}_{T1}$ [5]	QPSK	-46.5	-44.3
			$\underline{CH}_{T1}$ [6]	QPSK	-47	-44.8
			$\underline{CH}_{T1}$ [7]	QPSK	-47.5	-45.3
			$\underline{CH}_{T1}$ [8]	QPSK	-48	-45.8
			$\underline{CH}_{T1}$ [9]	QPSK	-48.5	-46.3
			$\underline{CH}_{T1}$ [10]	QPSK	-49	-46.8
			$\underline{CH}_{T1}$ [11]	QPSK	-49.5	-47.3
			$\underline{CH}_{T1}$ [12:24]	QPSK	-50	-47.8
			$\underline{CH}_{T2}$ [0:24]	QPSK	-44	-41.8
Hybrid	MA1	Reference Upper	CH_B	BPSK	-26	-23.8
		Reference Lower	CH_B	BPSK	-26	-23.8
		PIDS1	CH_{I1}	16-QAM	-43	-40.8
			CH_{I2}	16-QAM	-37	-34.8
		PIDS2	$CH_{PU} \cdot CH_{I3}$	16-QAM	-13 dB_{PU} (Note 3)	-13 dB_{PU} (Note 4)
			$CH_{PU} \cdot CH_{I4}$	16-QAM	-7 dB_{PU} (Note 3)	-7 dB_{PU} (Note 4)
			$CH_{PU} \cdot CH_{I5}$	16-QAM	0 dB_{PU} (Note 3)	0 dB_{PU} (Note 4)
		PIDS1*	CH_{I1}	16-QAM	-43	-40.8
			CH_{I2}	16-QAM	-37	-34.8
		PIDS2*	$CH_{PL} \cdot CH_{I3}$	16-QAM	-13 dB_{PL} (Note 5)	-13 dB_{PL} (Note 6)
			$CH_{PL} \cdot CH_{I4}$	16-QAM	-7 dB_{PL} (Note 5)	-7 dB_{PL} (Note 6)
			$CH_{PL} \cdot CH_{I5}$	16-QAM	0 dB_{PL} (Note 5)	0 dB_{PL} (Note 6)

Waveform	Service Mode	Sidebands	Amplitude Scale Factor Notation	Modulation Type	Nominal Power Spectral Density, dBc/Subcarrier	Nominal Power Spectral Density in a 300 Hz Bandwidth, dBc
All Digital	MA3	Primary Upper	CD_P	64-QAM	-15 (Note 7)	-12.8 (Note 7)
		Primary Lower	CD_P	64-QAM	-15 (Note 7)	-12.8 (Note 7)
		Secondary	CD_E	64-QAM	-30	-27.8
		Tertiary	CD_E	64-QAM	-30	-27.8
		Reference Upper	CD_B	BPSK	-15	-12.8
		Reference Lower	CD_B	BPSK	-15	-12.8
		PIDS1	$CD_P \cdot CH_{I1}$	16-QAM	-15 dB_{PU} (Note 8)	-15 dB_{PU} (Note 9)
			$CD_P \cdot CH_{I2}$	16-QAM	0 dB_{PU} (Note 8)	0 dB_{PU} (Note 9)
		PIDS2	$CD_P \cdot CH_{I1}$	16-QAM	-15 dB_{PL} (Note 10)	-15 dB_{PL} (Note 11)
			$CD_P \cdot CH_{I2}$	16-QAM	0 dB_{PL} (Note 10)	0 dB_{PL} (Note 11)

Notes:

1. In service mode MA1, the power spectral density of either the primary upper or primary lower sideband may be adjusted downward to any value provided that none of the specifications in Subsection 4.10 are violated.
2. In service mode MA1, only one primary sideband may be adjusted downward in power. The other sideband shall maintain its maximum power level.
3. The unit dB_{PU} refers to the power relative to the Primary Upper sideband. CH_{I3}, CH_{I4}, and CH_{I5} are adjusted so that the power spectral density (dBc per subcarrier) of the PIDS2 subcarrier has the value shown relative to the power spectral density of the Primary Upper sideband. For example, CH_{I3} is adjusted so that the power spectral density of the PIDS2 subcarrier is 13 dB below that of the Primary Upper sideband.
4. The unit dB_{PU} refers to the power relative to the Primary Upper sideband. CH_{I3}, CH_{I4}, and CH_{I5} are adjusted so that the power spectral density (in a 300-Hz bandwidth) of the PIDS2 subcarrier has the value shown relative to the power spectral density (in a 300-Hz bandwidth) of the Primary Upper sideband. For example, CH_{I3} is adjusted so that the power spectral density (in a 300-Hz bandwidth) of the PIDS2 subcarrier is 13 dB below that of the Primary Upper sideband.
5. The unit dB_{PL} refers to the power relative to the Primary Lower sideband. CH_{I3}, CH_{I4}, and CH_{I5} are adjusted so that the power spectral density (dBc per subcarrier) of the PIDS2* subcarrier has the value shown relative to the power spectral density of the Primary Lower sideband. For example, CH_{I3} is adjusted so that the power spectral density of the PIDS2* subcarrier is 13 dB below that of the Primary Lower sideband.

6. The unit dB_{PL} refers to the power relative to the Primary Lower sideband. CH_{I3}, CH_{I4}, and CH_{I5} are adjusted so that the power spectral density (in a 300-Hz bandwidth) of the PIDS2* subcarrier has the value shown relative to the power spectral density (in a 300-Hz bandwidth) of the Primary Lower sideband. For example, CHI3 is adjusted so that the power spectral density (in a 300-Hz bandwidth) of the PIDS2* subcarrier is 13 dB below that of the Primary Lower sideband.
7. In service mode MA3, both primary sidebands shall be set to the maximum power spectral density values shown. The level is not adjustable.
8. The unit dB_{PU} refers to the power relative to the Primary Upper sideband. CD_{I1} and CD_{I2} are adjusted so that the power spectral density (dBc per subcarrier) of the PIDS1 subcarrier has the value shown relative to the power spectral density of the Primary Upper sideband. For example, CD_{I1} is adjusted so that the power spectral density of the PIDS1 subcarrier is 15 dB below that of the Primary Upper sideband.
9. The unit dB_{PU} refers to the power relative to the Primary Upper sideband. CD_{I1} and CD*I2* are adjusted so that the power spectral density (in a 300-Hz bandwidth) of the PIDS1 subcarrier has the value shown relative to the power spectral density (in a 300-Hz bandwidth) of the Primary Upper sideband. For example, CD_{I1} is adjusted so that the power spectral density (in a 300-Hz bandwidth) of the PIDS1 subcarrier is 15 dB below that of the Primary Upper sideband.
10. The unit dB_{PL} refers to the power relative to the Primary Lower sideband. CD_{I1} and CD_{I2} are adjusted so that the power spectral density (dBc per subcarrier) of the PIDS2 subcarrier has the value shown relative to the power spectral density of the Primary Lower sideband. For example, CD_{I1} is adjusted so that the power spectral density of the PIDS2 subcarrier is 15 dB below that of the Primary Lower sideband.
11. The unit dB_{PL} refers to the power relative to the Primary Lower sideband. CD_{I1} and CD_{I2} are adjusted so that the power spectral density (in a 300-Hz bandwidth) of the PIDS2 subcarrier has the value shown relative to the power spectral density (in a 300-Hz bandwidth) of the Primary Lower sideband. For example, CD_{I1} is adjusted so that the power spectral density (in a 300-Hz bandwidth) of the PIDS2 subcarrier is 15 dB below that of the Primary Lower sideband.

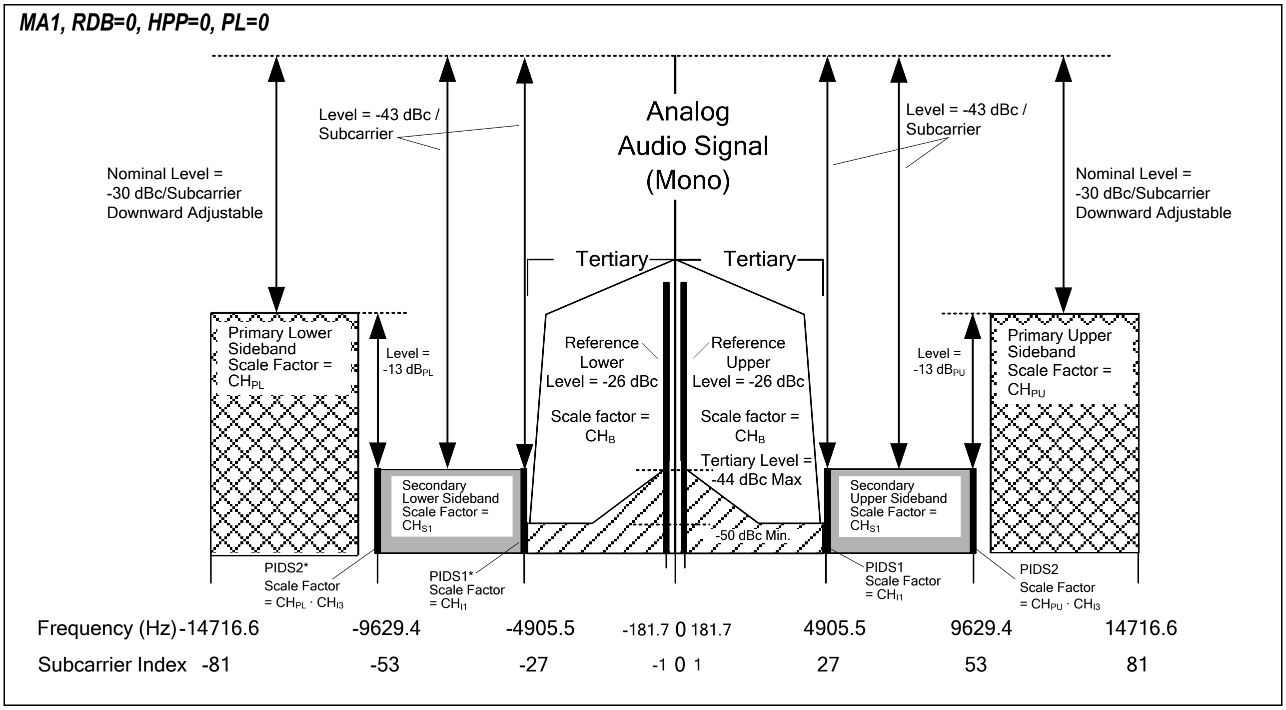

Figure 4-6: MA1, RDB=0, HPP=0, PL-0 – Symmetrical Sidebands

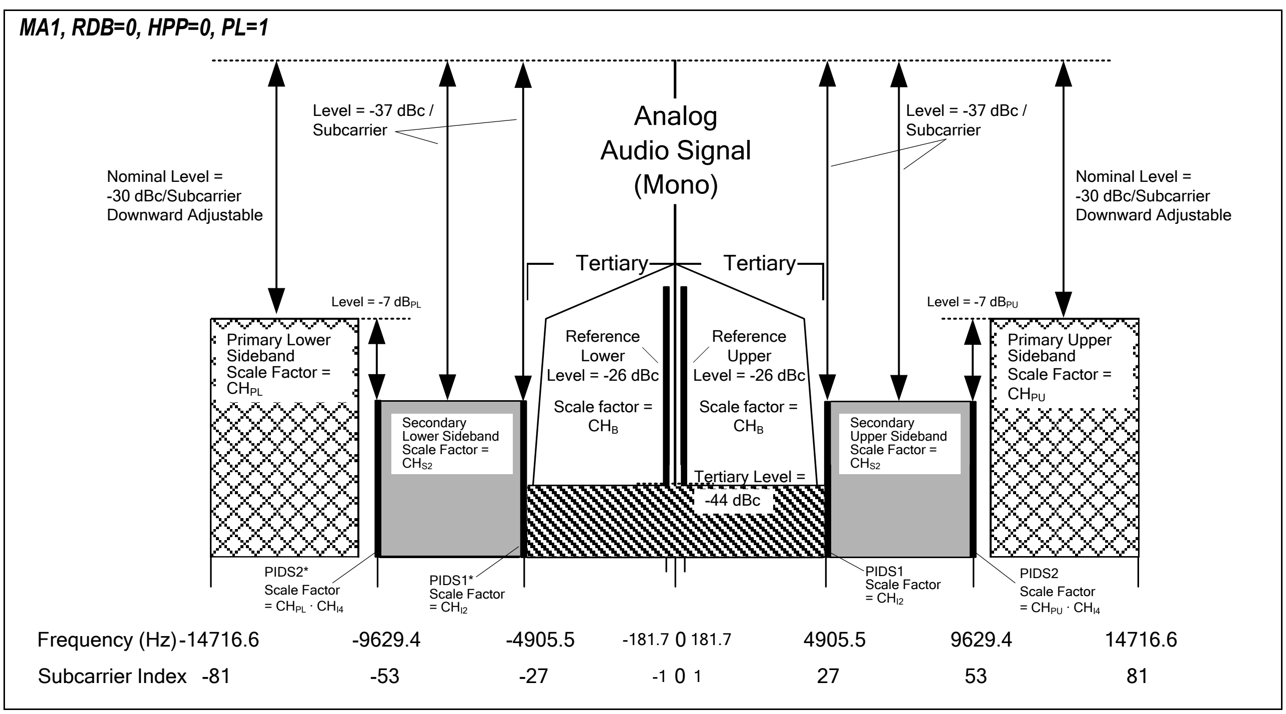

Figure 4-7: MA1, RDB=0, HPP=0, PL=1 – Symmetrical Sidebands

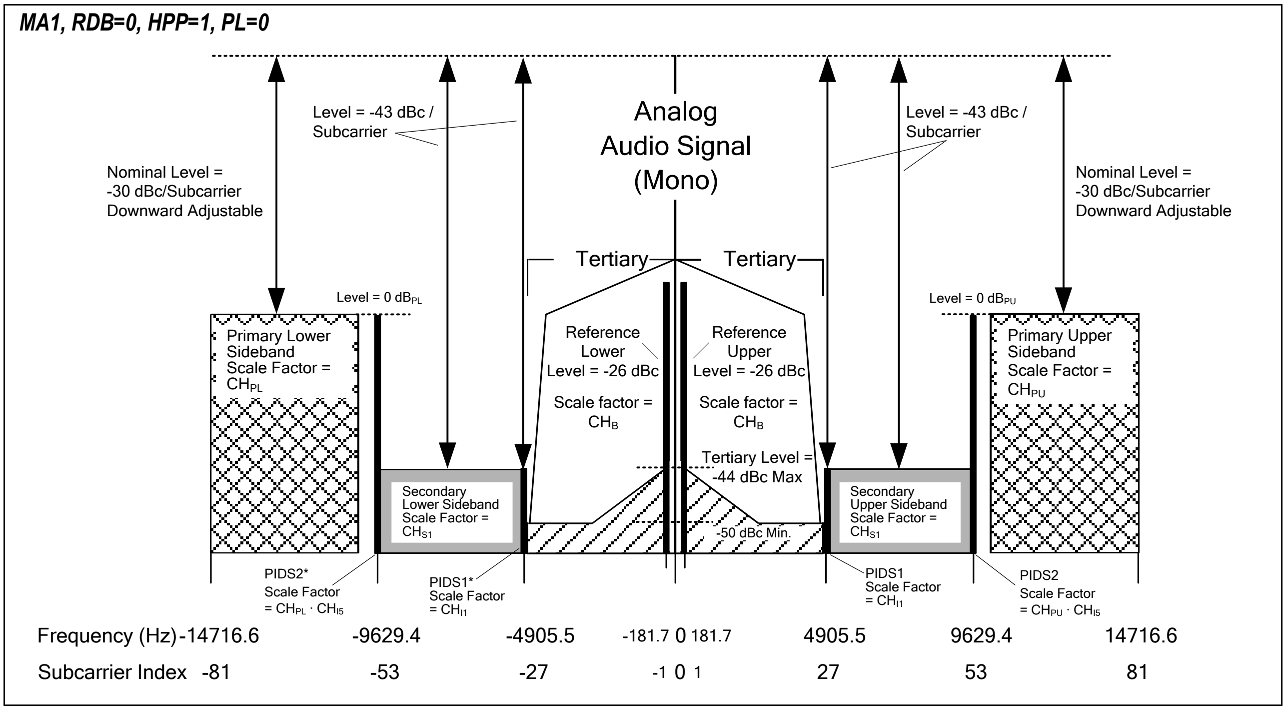

Figure 4-8: MA1, RDB=0, HPP=1, PL=0 – Symmetrical Sidebands

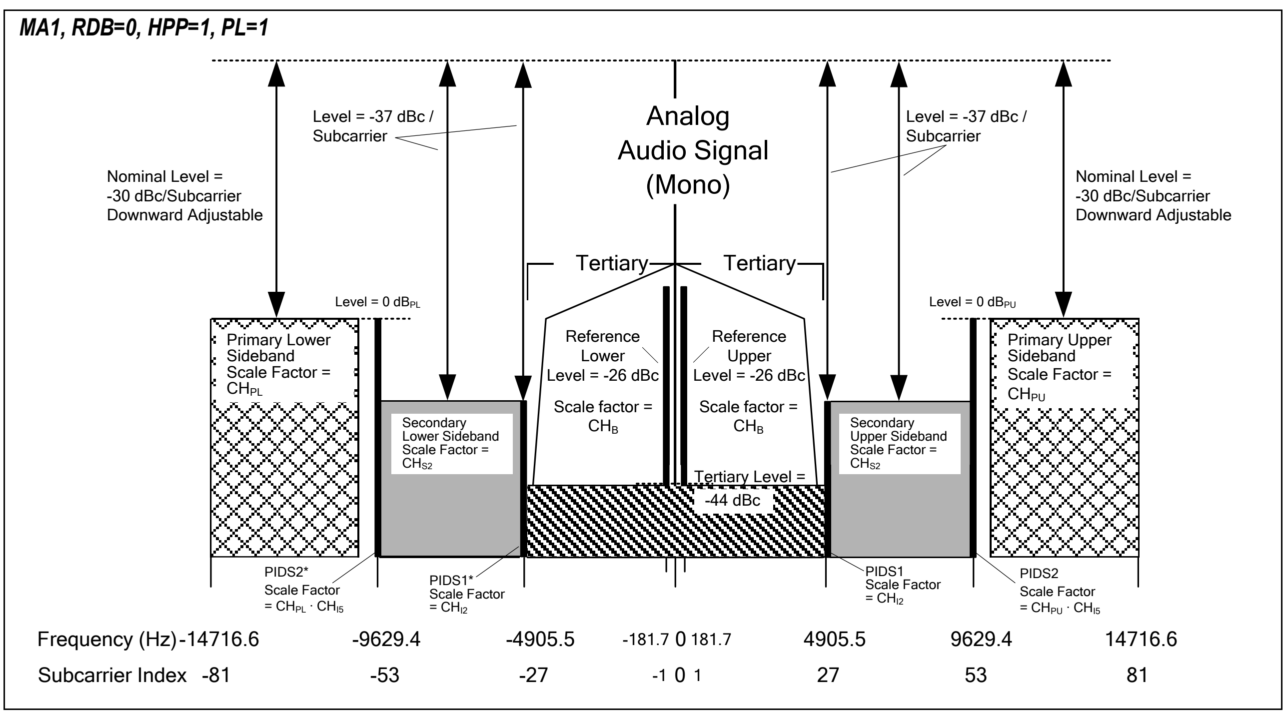

Figure 4-9: MA1, RDB=0, HPP=1, PL=1 – Symmetrical Sidebands

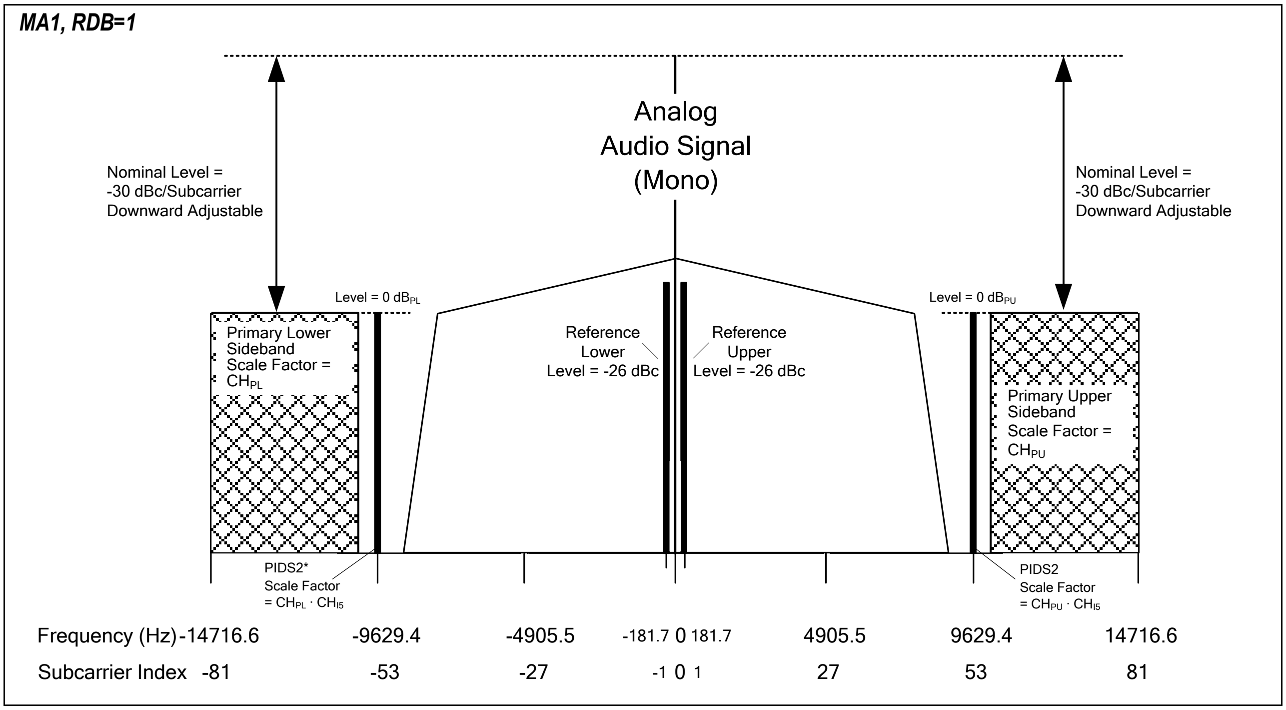

Figure 4-10: MA1, RDB=1 – Symmetrical Sidebands

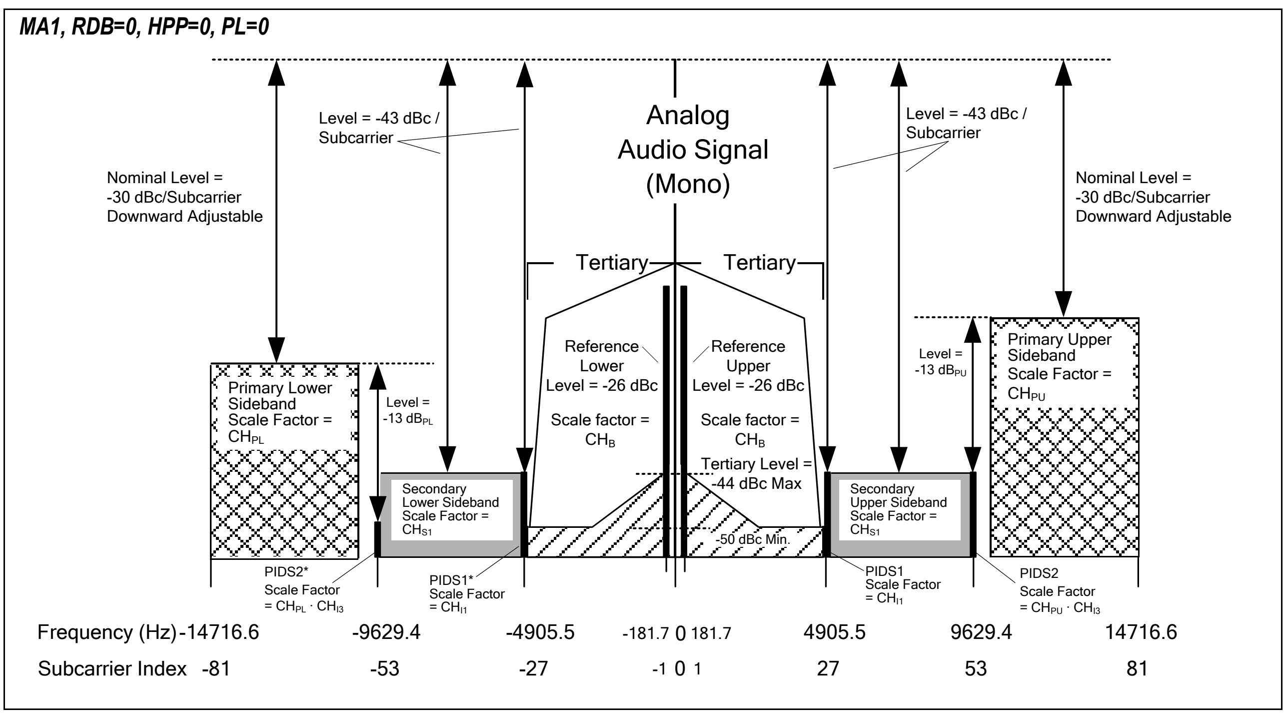

Figure 4-11: MA1, RDB=0, HPP=0, PL-0 – Asymmetrical Sidebands

NOTE

In Figure 4-11, the Primary Upper and Lower Sidebands are shown to have different power levels to illustrate an asymmetrical sideband configuration. Normally the two sidebands are equal but may be different under special operational scenarios.

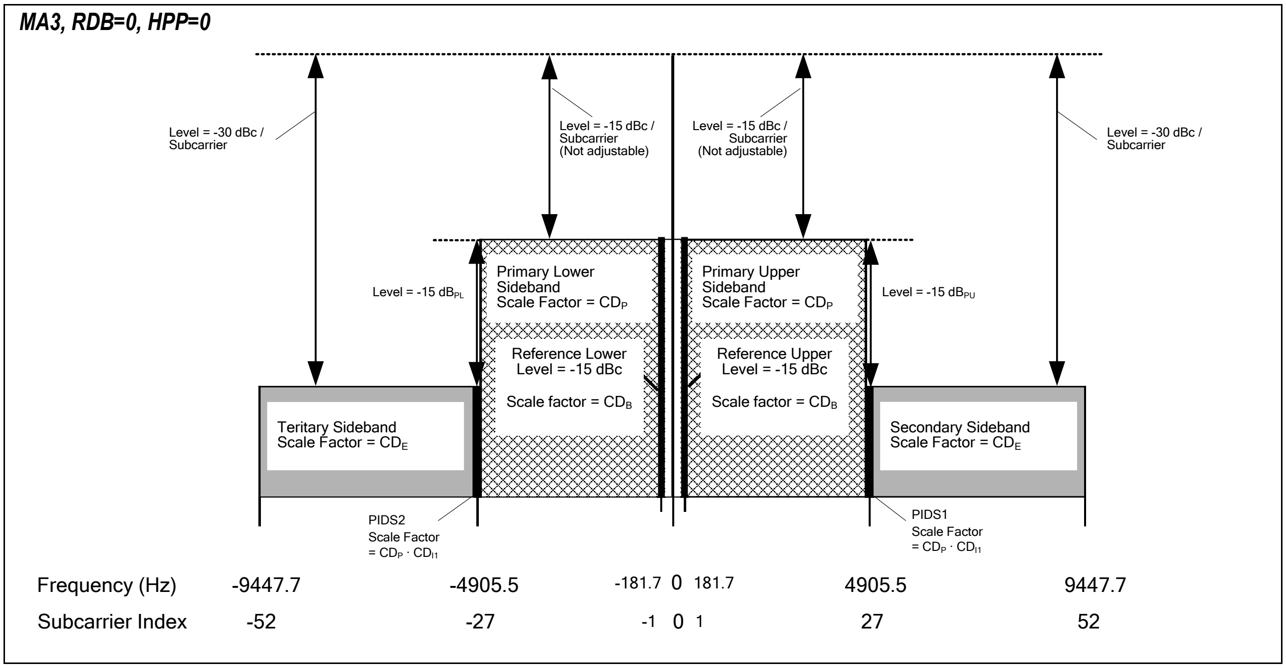

Figure 4-12: MA3, RDB=0, HPP=0

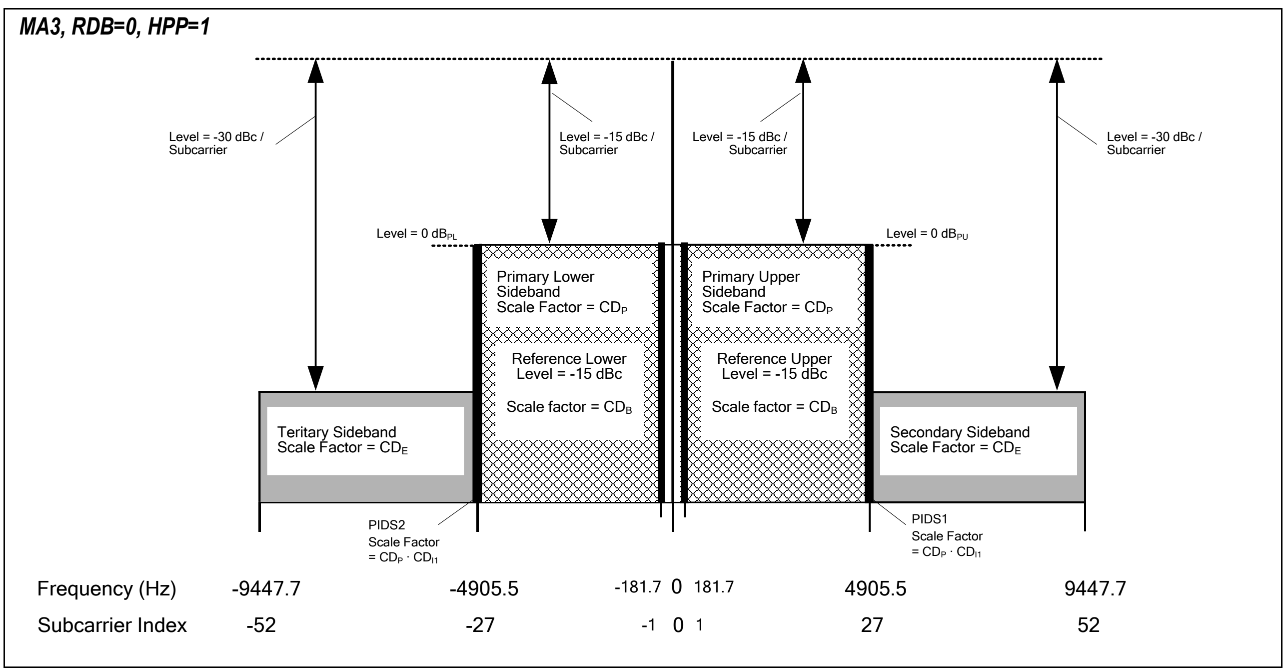

Figure 4-13: MA3, RDB=0, HPP=1

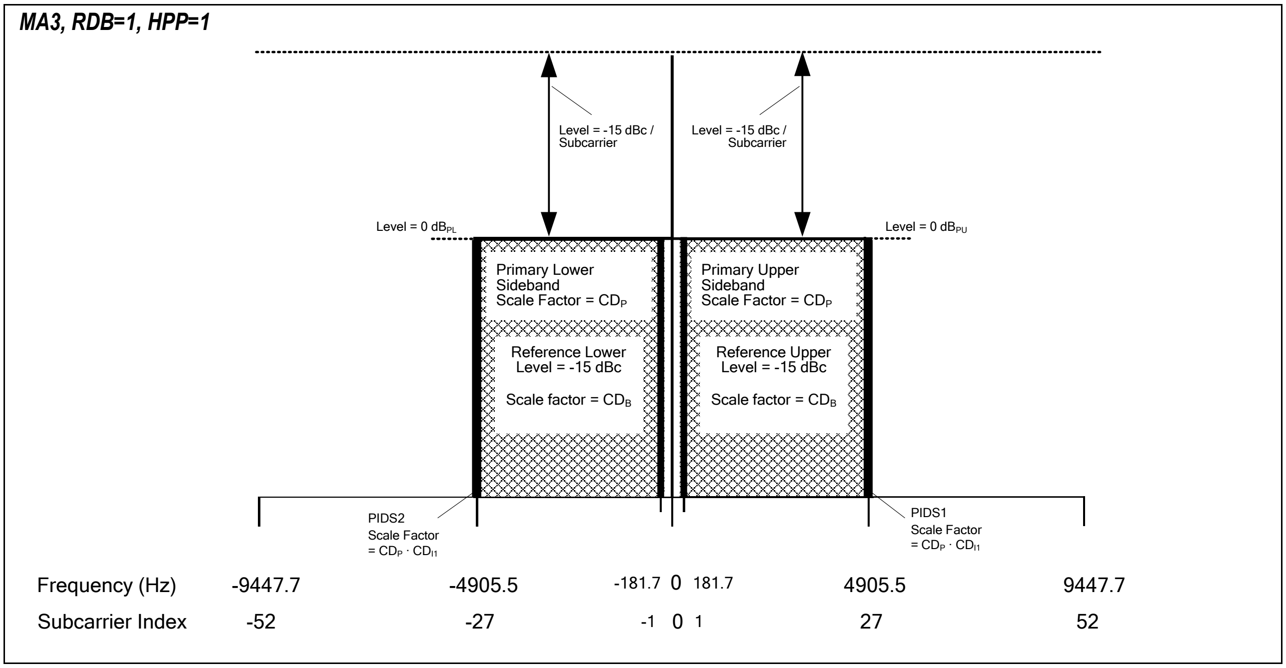

Figure 4-14: MA3, RDB=1, HPP=1

4.6.1 AM Digital Carrier Power

4.6.1.1 Hybrid MA1 Digital Carrier Power

Table 4-7 characterizes the total (nominal) integrated digital power for the various Hybrid MA1 waveform configurations. The nominal digital-to-analog power ratio is derived from Table 4-6, where 0 dBc equals the total power of the unmodulated analog AM carrier.

In addition, the total integrated digital power with one of the primary digital sidebands removed is shown. This represents a lower power limit for asymmetric sideband operation (for calculation purposes only; it is not expected that a sideband will be completely shut off). This value, in combination with the total digital power of just one of the primary sidebands alone is useful to calculate the exact power level for asymmetric sideband operation, as explained in the next subsection.

Table 4-7: Hybrid MA1 Digital to Analog Power Ratios

Subcarrier Scaling Control Signal State			Total Digital Power of All Sidebands (Nominal)	Total Digital Power with One Primary Sideband Removed (Nominal)	Total Digital Power of One Primary Sideband Alone (Nominal)
RDB	HPP	PL			
0	0	0	-12.33 dBc	-14.74 dBc	-16.02 dBc
0	0	1	-11.69 dBc	-13.69 dBc	-16.02 dBc
0	1	0	-12.19 dBc	-14.51 dBc	-16.02 dBc
0	1	1	-11.59 dBc	-13.53 dBc	-16.02 dBc
1	X	X	-12.44 dBc	-14.95 dBc	-16.02 dBc

4.6.1.2 Power Limits for MA1 Asymmetrical Sideband Operation

If asymmetrical sideband operation is desired, the last two columns of Table 4-7 can be used to help calculate the total integrated power, but the total digital power of one primary sideband is reduced by the desired power reduction in dB. For example, for RDB=0, HPP=0, and PL=0 and it is desired to reduce the Primary Lower sideband power level by 12 dB, the total integrated power will be:

$= 10 \text{ Log}_{10} (\text{Log}_{10}{}^{-1} (\text{PWR1} /10) + \text{Log}_{10}{}^{-1} ([\text{Pwr2} – \text{Sideband Reduction Value}] /10))$

Where

PWR1 = Total Digital Power with One Primary Sideband Removed

PWR2 = Total Digital Power of One Primary Sideband Alone

$= 10 \text{ Log}_{10} (\text{Log}_{10}{}^{-1} (-14.74 /10) + \text{Log}_{10}{}^{-1} ([-16.02 – 12] /10))$

$= 10 \text{ Log}_{10} (0.03357 + 0.001578)$

$= 10 \text{ Log}_{10} (0.03515)$

$= -14.54$ dBc

4.6.1.3 All Digital MA3 Carrier Power

Table 4-8 characterizes the total integrated digital for the various Hybrid MA3 waveform configurations. The nominal digital-to-unmodulated AM carrier power ratio is derived from Table 4-6, where 0 dBc equals the total power of the unmodulated analog AM carrier.

Table 4-8: All Digital MA3 Carrier Power

Subcarrier Scaling Control Signal State			Total Digital Power of All Sidebands
RDB	HPP	PL	
0	0	X	+2.30 dBc
0	1	X	+2.45 dBc
1	X	X	+2.32 dBc

4.7 Analog Audio Source

The analog signal shall not exceed the modulation levels specified in Title 47 CFR §73.1570.

The HD Radio system is not compatible with existing analog AM stereophonic broadcasts. The input analog signal shall be a monophonic signal.

4.8 Phase Noise

The phase noise mask for the broadcast system is illustrated in Figure 4-15 and Figure 4-16 and specified in Table 4-9. As can be seen in the figures, the response is linear (on the dB scale) between every pair of points drawn on the curve.

Zero dBc is defined as the total power of the subcarrier being measured. The phase noise mask is applicable for all permissible power levels of the upper and lower sidebands, as defined in Subsection 4.6.

The total single sideband phase noise at the transmitter RF output as measured in a 1-Hz bandwidth shall be within the mask specified in Table 4-9. This shall be verified by transmitting a single unmodulated digital subcarrier. In addition, for the Hybrid waveform, the unmodulated AM carrier shall be separately verified.

Table 4-9: AM Broadcast System Phase Noise Specifications

Frequency, F, Offset Relative to Carrier	Level, dBc/Hz
1 Hz to 10 Hz	$-1.11F - 38.9$
10 Hz to 100 Hz	$-4.44 \cdot 10^{-1}F - 45.6$
100 Hz to 1000 Hz	$-5.56 \cdot 10^{-3}F - 89.4$
1 kHz to 10 kHz	$-1.67 \cdot 10^{-3}F - 93.3$
10 kHz to 100 kHz	$-1.11 \cdot 10^{-4}F - 108.9$
> 100 kHz	-120.0

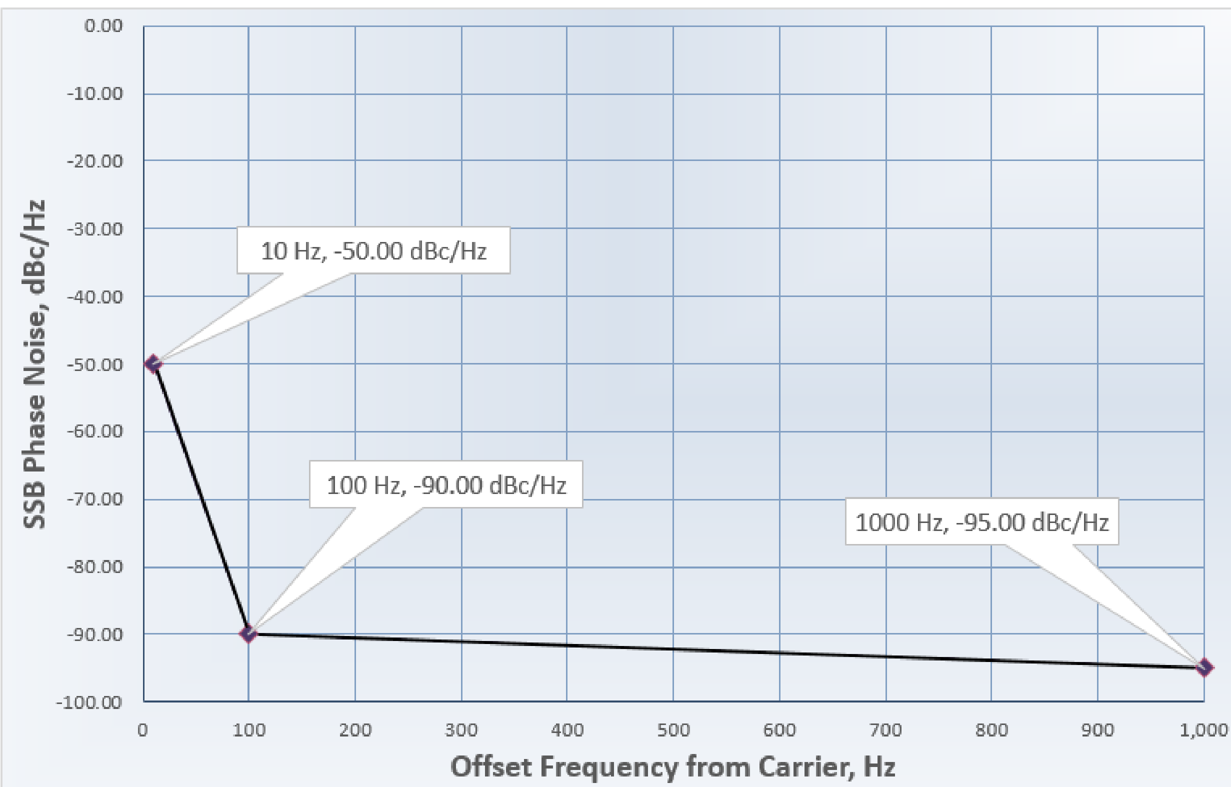

Figure 4-15: AM SSB Phase Noise Mask | 10 Hz to 1000 Hz

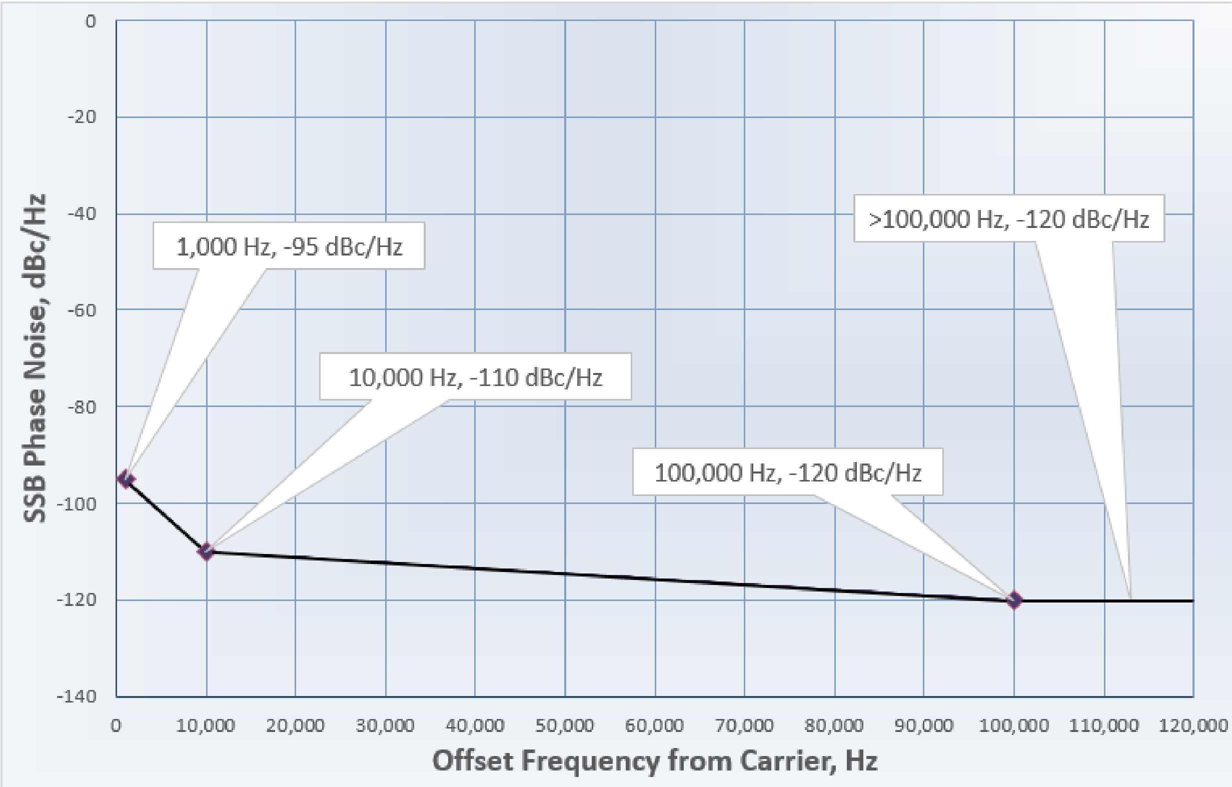

Figure 4-16 : AM SSB Phase Noise Mask | 1 kHz to 100 kHz

4.9 Discrete Phase Noise

For the broadcast system, the spectrum from (F_c – 15 kHz) to (F_c + 15 kHz) shall be considered to consist of multiple non-overlapping sub-bands, each with a bandwidth of 100 Hz, where F_c is the carrier frequency. Discrete phase noise components measured at the transmitter RF output shall be permitted to exceed the mask specified in Table 4-6 provided that for each sub-band, the measured total integrated phase noise does not exceed the total integrated phase noise calculated from Table 4-9.

If the upper and lower sidebands have different power levels, as permitted in Subsection 4.6, the measurement must account for the fact that the 0-dBc reference level will be different for each sideband.

4.10 Modulation Error Ratio (MER)

AM MER is defined as the ratio of the signal power to the error power as defined in Equation 1. The ratio shall be computed for each subcarrier type independently as explained in the following subsections.

Equation 1: Classical MER Computation

$$\mathrm{MER(dB)} = 10\log_{10}\left(\frac{P_{\mathbf{signal}}}{P_{\mathbf{error}}}\right)$$

where P_{error} is the RMS power of the error vector, and P_{signal} is the RMS power of the ideal transmitted signal

4.10.1 Detailed MER Specifications

The following specifications shall be met, using the test configuration described in Reference [26].

4.10.1.1 MA1 and MA3 Reference Subcarriers

The instantaneous MER for each and every Binary Phase Shift Keying (BPSK) reference subcarrier, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 11 dB, as computed by Equation 1.

The average MER of all the Binary Phase Shift Keying (BPSK) reference subcarrier in the upper sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 14 dB, as computed by Equation 1. This computation shall be based on a block of N = 128 contiguous symbols.

The average MER of all the Binary Phase Shift Keying (BPSK) reference subcarrier in the lower sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 14 dB, as computed by Equation 1. This computation shall be based on a block of N = 128 contiguous symbols.

4.10.1.2 MA1 Tertiary Subcarriers

The instantaneous MER for each and every Quadrature Phase Shift Keying (QPSK) subcarrier, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 11 dB, as computed by Equation 1.

The average MER of all the Quadrature Phase Shift Keying (QPSK) subcarriers in the upper sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 14 dB, as computed by Equation 1, averaged across all tertiary upper QPSK subcarriers. This computation shall be based on a block of N = 128 contiguous symbols.

The average MER of all the Quadrature Phase Shift Keying (QPSK) subcarriers in the lower sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 14 dB, as computed by Equation 1, averaged across all tertiary lower QPSK subcarriers. This computation shall be based on a block of N = 128 contiguous symbols.

4.10.1.3 MA1 Secondary, MA1 PIDS, and MA3 PIDS Subcarriers

The instantaneous MER for each and every 16-QAM signal subcarrier, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 18 dB, as computed by Equation 1.

The average MER of all the 16-QAM subcarriers in the upper sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 21 dB, as computed by Equation 1, averaged across all upper sideband 16-QAM subcarriers. This computation shall be based on a block of N = 128 contiguous symbols.

The average MER of all the 16-QAM subcarriers in the lower sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 21 dB, as computed by Equation 1, averaged across all lower sideband 16-QAM subcarriers. This computation shall be based on a block of N = 128 contiguous symbols.

4.10.1.4 MA3 Secondary and Tertiary Subcarriers

The instantaneous MER for each and every MA3 Secondary and Tertiary 64-QAM signal subcarrier, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 22 dB, as computed by Equation 1.

The average MER of all the MA3 Secondary and Tertiary 64-QAM subcarriers in the upper sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 25 dB, as computed by Equation 1, averaged across all upper sideband 64-QAM subcarriers. This computation shall be based on a block of N = 128 contiguous symbols.

The average MER of all the MA3 Secondary and Tertiary 64-QAM subcarriers in the lower sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 25 dB, as computed by Equation 1, averaged across all lower sideband 64-QAM subcarriers. This computation shall be based on a block of N = 128 contiguous symbols.

4.10.1.5 MA1 Primary Subcarriers

The instantaneous MER for each and every MA1 Primary 64-QAM signal subcarrier, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 22 dB, as computed by Equation 1.

The average MER of all the MA1 Primary 64-QAM subcarriers in the upper sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 25 dB, as computed by Equation 1, averaged across all upper sideband 64-QAM subcarriers. This computation shall be based on a block of N = 128 contiguous symbols.

The average MER of all the MA1 Primary 64-QAM subcarriers in the lower sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 25 dB, as computed by Equation 1, averaged across all lower sideband 64-QAM subcarriers. This computation shall be based on a block of N = 128 contiguous symbols.

4.10.1.6 MA3 Primary Subcarriers

The instantaneous MER for each and every MA3 Primary 64-QAM signal subcarrier, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 22 dB, as computed by Equation 1.

The average MER of all the MA3 Primary 64-QAM subcarriers in the upper sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 25 dB, as computed by Equation 1, averaged across all upper sideband 64-QAM subcarriers. This computation shall be based on a block of N = 128 contiguous symbols.

The average MER of all the MA3 Primary 64-QAM subcarriers in the lower sideband, measured at the RF output of the transmission system at the connection point to the antenna system (including any RF filters), shall be greater than or equal to 25 dB, as computed by Equation 1, averaged across all lower sideband 64-QAM subcarriers. This computation shall be based on a block of N = 128 contiguous symbols.

4.11 Gain Flatness

The total gain of the transmission signal path as verified at the transmitter output into a 50-ohm, non-reactive load, shall be flat to within ±0.5 dB for all frequencies between (F_c – 10 kHz) to (F_c + 10 kHz), where F_c is the RF channel frequency. For frequencies removed from F_c by more than 10 kHz and less than 15 kHz, the gain shall be flat to within ±1.0 dB. It is assumed that the source data consists of scrambled binary ones and the power of each subcarrier is an average value.

For the case where the upper and lower digital sideband power levels are intended to be different, as defined in Subsection 4.6, the gain flatness specification shall be interpreted as follows:

> Gain flatness is the difference between the measured power spectral density in a 300-Hz bandwidth of each subcarrier frequency, and the power spectral density of the applicable digital sideband, normalized to a 300-Hz bandwidth.

For optimal HD Radio digital performance it is recommended that the transmission system, including the antenna, adheres as closely as is practicable to the Gain Flatness specification. Performance may be verified using a suitable sample loop on the reference or main tower. In addition to antenna component selection and adjustment, active pre-compensation of the HD Radio waveform may be employed to improve the effective gain flatness.

4.12 Amplitude and Phase Symmetry

The amplitude and phase symmetry of the transmission signal path shall be verified at the transmitter output into a 50-ohm, non-reactive load. For Hybrid transmissions, for any frequency, F, between 0 and 5 kHz, removed from the carrier frequency, F_c, the RF digital transmission must maintain symmetry within the following limits:

i. The average RF signal power at a frequency (F_c + F) shall be within ±0.25 dB of the RF signal power at the corresponding frequency (F_c – F), where the power is measured in a 300-Hz bandwidth averaged over an interval of at least 30 seconds of time and for at least 100 averages.

ii. The phase of the signal at a frequency (F_c + F) shall be equal to the negative of the signal phase at a frequency (F_c – F) within ±2 degrees rms.

For optimal HD Radio digital performance it is recommended that the transmission system, including the antenna, adheres as closely as is practicable to the Amplitude and Phase Symmetry specification. This may be verified using a suitable sample loop on the reference or main tower. In addition to antenna component selection and adjustment, active pre-compensation of the HD Radio waveform may be employed to improve the amplitude and phase symmetry.

The above specifications assume that the upper and lower digital sidebands are transmitted with equal power levels. If this is not the case, the appropriate power level offset shall be applied to the amplitude symmetry specification listed above.

4.13 Group Delay Flatness

Group delay of the transmission signal path shall be verified at the transmitter output into a 50-ohm, non-reactive load. The group delay of the entire transmission signal path (excluding the RF channel) as measured at the RF channel frequency (F_c) shall be flat to within ±3 µs from (F_c – 15 kHz) to (F_c + 15 kHz).

For optimal HD Radio digital performance it is recommended that the transmission system, including the antenna, adheres as closely as is practicable to the Group Delay specification. This may be verified using a suitable sample loop on the reference or main tower. In addition to antenna component selection and adjustment, active pre-compensation of the HD Radio waveform may be employed to improve group delay.

HD Radio™ Air Interface Design Description Program Service Data Transport

Rev. D

December 14, 2016

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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 defines the Program Service Data (PSD) transport. It describes the PSD packet encapsulation process for use by the audio transport. A specific hardware and software implementation is not described.

2 Referenced Documents

STATEMENT
Each referenced document that is mentioned in this document shall be listed in the following iBiquity document:

- Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard
  Document Number: SY_REF_2690s

3 Abbreviations and Conventions

3.1 Abbreviations and Acronyms

CRC	Cyclic Redundancy Check
FCS	Frame Check Sequence
HDLC	High Level Data Link Control
IETF	Internet Engineering Task Force
LCP	Link Control Protocol
OTA	Over the Air
PPP	Point-to-Point Protocol
PSD	Program Service Data
PDU	Protocol Data Unit
RFC	Request For Comments
SEQ	Sequence Number

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.
- In little-endian format, a multi-byte value is stored in memory from the lowest byte (the "little end") to the highest byte. For example, all 2-byte messages are little-endian where the upper and lower bytes are swapped.

4 Overview

4.1 Introduction

Program Service Data (PSD) consists of ID3 tags which are transmitted as data packets [8]. The PSD transport provides the packet transport mechanism for the PSD. Data packets are encapsulated as a byte stream for transmission and recovery at the receiver. The encapsulated packets, referred to as PSD PDUs (Protocol Data Units), are transmitted within the encoded audio stream. The packet transport link for the HD Radio system is described in Section 5. On the receiver side, packets are recovered from the encoded audio stream and the ID3 payloads are made available for use on the receiver.

Figure 4-1 shows the transport of PSD (ID3 tags) from the transmit side to the receive side.

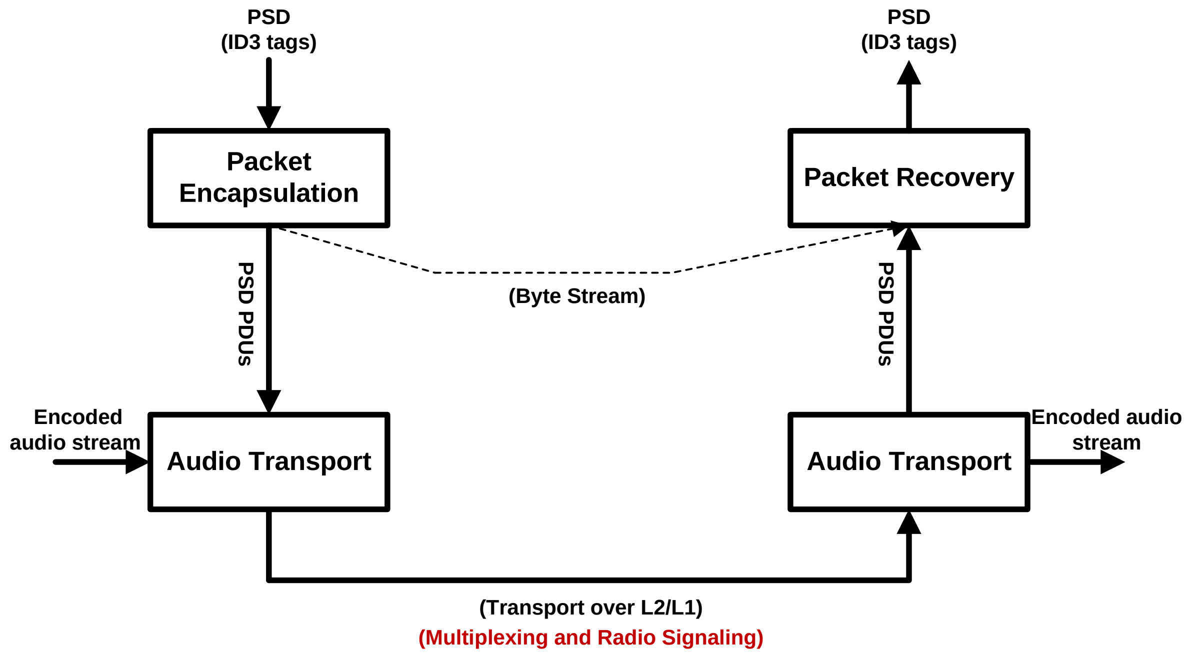

Figure 4-1: Program Service Data Transport – High Level View

4.2 Audio Transport Protocol

The Audio Transport obtains the PSD byte streams, if present, and multiplexes them with the encoded audio packets. The PSD streams are continually repeated throughout the transmission. Once the buffer is filled with content, the system repeatedly transmits the content until the buffers are updated. The transmission rate depends on the available bandwidth and the length of the message. This multiplexing of each PSD byte within the audio stream is handled internally by the Audio Transport. Thus, the output streams from the Audio Transport contain both the compressed audio and the PSD byte streams. Refer to [4] for a detailed description of the Audio Transport.

4.3 ID3 Tag Generation

The ID3 tags are generated from the Service Interface and provided along with the audio content. Refer to [8] for a detailed definition of the use of ID3 tags in the HD Radio system and the ID3 tag fields supported by the HD Radio system.

4.4 Packet Transport

Packet transport for the PSD is provided by the PSD Transport. At a high level, the PSD Transport receives an input packet from a higher layer, translates it internally to an over-the-air (OTA) packet in a robust and efficient manner to its peer at a receiver, where it is translated to an output packet and sent to higher level peer.

The PSD Transport also supplies the functions of packet encapsulation (at the transmitter) and packet recovery (at the receiver). In addition to payload transport, the PSD packet structure also provides:

- Addressing schemes to allow association of packets to services.
- Sequence control to assure packet order preservation and detection.
- Error detection information to allow reliable packet detection.

The PSD Packet Transport is described in Section 5.

5 Program Service Data Transport

The PSD Transport provides a generic and reliable packet transport for implementing data services. This section describes the PSD Transport and PSD PDU generation.

5.1 Packet Encapsulation – PDU Generation

The packet encapsulation used by the PSD Transport follows the HDLC-like framing employed by the Point-to-Point Protocol (PPP) as standardized by the IETF in RFC 1662 [24]. The following sections describe how the HDLC-like framing of PPP has been adapted for the HD Radio system.

The HDLC-like framing allows encapsulation of a packet within a byte stream, referred to as PSD PDU, that may be sent in segments of arbitrary size (e.g., in each modem frame). Reconstruction of the packet requires only concatenation of the segments. Depending on their size, a single modem frame may contain multiple such encapsulated packets or a single portion of a large packet.

A modem frame refers to a Layer 1 frame base on the size of the logical channels as defined in [1] and [2].

5.1.1 PDU Format

A PSD PDU is contained in an HDLC-like frame delimited by Flags as shown in Table 5-1.

Table 5-1: PSD PDU Field Definition

Field	Bytes	Description
Flag	1	0x7E (Start of PDU)
Protocol Field	1	Protocol Field = 0x21 for the PSD packet format
Information	As required	PSD packets as defined in Subsection 5.2
FCS	2	A 16-bit Frame Check Sequence is used for error detection In little-endian format
Flag	1	0x7E (Start of next PDU)

This frame structure follows that described in RFC 1662, Section 3.1 [24] except for the following changes:

1. The Address and Control fields provide no useful function in the HD Radio system and have been eliminated in the interest of efficiency.
2. The Protocol Field is always 8-bits and has a value less than 0x80 (greater values are reserved for future expansion).
3. No padding is used.
4. The Frame Check Sequence is always 16-bits for Protocol Fields complying with item 2.

5.1.1.1 Flag Delimiters

Each HDLC-like frame is delimited by Flag bytes having the value 0x7E. The Flag delimiters serve the following purposes:

- Only one byte is needed to delimit a packet of any length.
- A false Flag due to a payload error results only in the loss of a single frame (packet).
- A corrupted Flag cannot cause a loss of more than two frames (packets).

A single modem frame may consist of partial or multiple instances of such HDLC-like frames. The flag bytes help in identifying and delimiting each frame in such instances.

5.1.1.2 Protocol Field

The Protocol Field is used to allow for multiple packet formats to be supported. For PSD, the default protocol (0x21) is used.

On receipt, any frame with an unrecognized Protocol Field should be discarded by the Audio Transport. This allows new packet protocols to be added in the future while retaining backward compatibility with older receivers.

5.1.1.3 Information

The Information field contains PSD packets as defined in Subsection 5.2.

5.1.1.4 Frame Check Sequence

The Frame Check Sequence (FCS) uses a 16-bit CRC. The FCS is generated using the Protocol Field and the Information data (refer to Table 5-2) in accordance with RFC 1662, Section C.1 [24]. Refer to [24] for a definition for FCS. The FCS is used in little-endian format.

5.1.2 Transparency

To prevent a value of 0x7E occurring in the data from being read as a Flag, an escape mechanism is provided to replace bytes with a special meaning with alternate values. This is done by replacing the byte with two bytes consisting of the control escape byte 0x7D followed by the original byte exclusive-or'ed with hexadecimal 0x20 (Refer to RFC 1662, Section 4.2 [24]). The only two values that need to be escaped are:

0x7E which is encoded as 0x7D, 0x5E, (Flag Sequence)

0x7D which is encoded as 0x7D, 0x5D (Control Escape)

Since the escape mechanism requires two bytes to encode a single byte, it reduces efficiency slightly: about 1% for a packet with a random data payload. Since the Flag and Control Escape characters correspond to the characters "~" and "}", which seldom appear in ID3 data, the efficiency loss on PSD packets is much less than 1%.

5.1.3 Idle Pattern

When no packet data is available to send, an idle pattern of repeating Flags is sent. This is equivalent to a stream of zero length frames.

5.1.4 Application of RFC-1662 for the HD Radio System

Many of the features defined in RFC 1662 are inapplicable or unnecessary for the HD Radio system. In particular, the following sections of the RFC 1662 are not applicable to the HD Radio system:

- Sections 4.4.2, 4.5.2, and 5 – All streams used for packet transport in the HD Radio system are octet-synchronous.
- Section 6 – No asynchronous to synchronous conversion is used.
- Section 7 – The Flag Sequence and Control Escape are the only control flags used in the HD Radio system. Negotiation of additional control characters is not possible and not required.
- Section A – LCP negotiations are not possible and are not used.
- Section B – The PPP frames identified are not valid frames in the HD Radio system.

5.2 Default PSD Packet Definition

Program Service Data uses the default packet format shown in Table 5-2.

Table 5-2: Default Packet Definition

Field	Size (bytes)	Description
Port	2	Port number for addressing a particular service In little-endian format
SEQ	2	Sequence number increments by 1 on each packet sent In little-endian format.
Payload[ ]	1 to 1024	Payload length is variable up to 1024 bytes

The following subsections describe the corresponding fields and how they are used for PSD.

5.2.1 Port Number

Port numbers are used to allow packets to be directed to specific applications. Port number 0x5100 is used for Main Program Service Data and 0x5201 through 0x5207 are reserved for future PSD applications. The Port number is used in little-endian format.

5.2.2 Sequence Number

At the transmitter, each packet sent to a given PORT has a sequence number one greater than the previous one. This allows for packet order to be verified at the receiver and for lost packets to be detected through missing sequence numbers. The sequence number is used in little-endian format.

5.2.3 Packet Payload

The packet payloads are of variable length up to 1024 bytes. For PSD, the payload data is an ID3 tag. Large packets may be transmitted over multiple modem frames. Please refer to [4] for the MPS Data rates.

5.3 PSD PDU - Example

Figure 5-1 shows an example of a PSD PDU. The following elements are noted in the figure:

1. The beginning of the frame indicated by a Flag Sequence (0x7E).
2. The first byte of the frame is the Protocol ID field which is set to 0x21, indicating the default packet format.
3. The next two bytes contain the Port number, 0x5100, in little-endian format.
4. A two-byte Sequence Number in little-endian format follows next. The value of 0x0000 is meaningful only with respect to the sequence numbers of the previous and subsequent packets sent to port 0x5100.
5. The payload is an ID3 tag that encodes the song title (“Analog Blues”), the artist (“J. Q. Public”), and the album name (“The Lost Sessions”).
6. The payload is followed by a two-byte Frame Check Sequence in little-endian format. It is computed over all bytes from the Protocol ID field through the last byte of the payload.
7. The end of frame is indicated by a Flag Sequence (0x7E).

NOTE
The byte stream shown could arrive as segments of arbitrary size so long as the byte order is preserved. Lost segments will result in short packets that fail FCS checking.

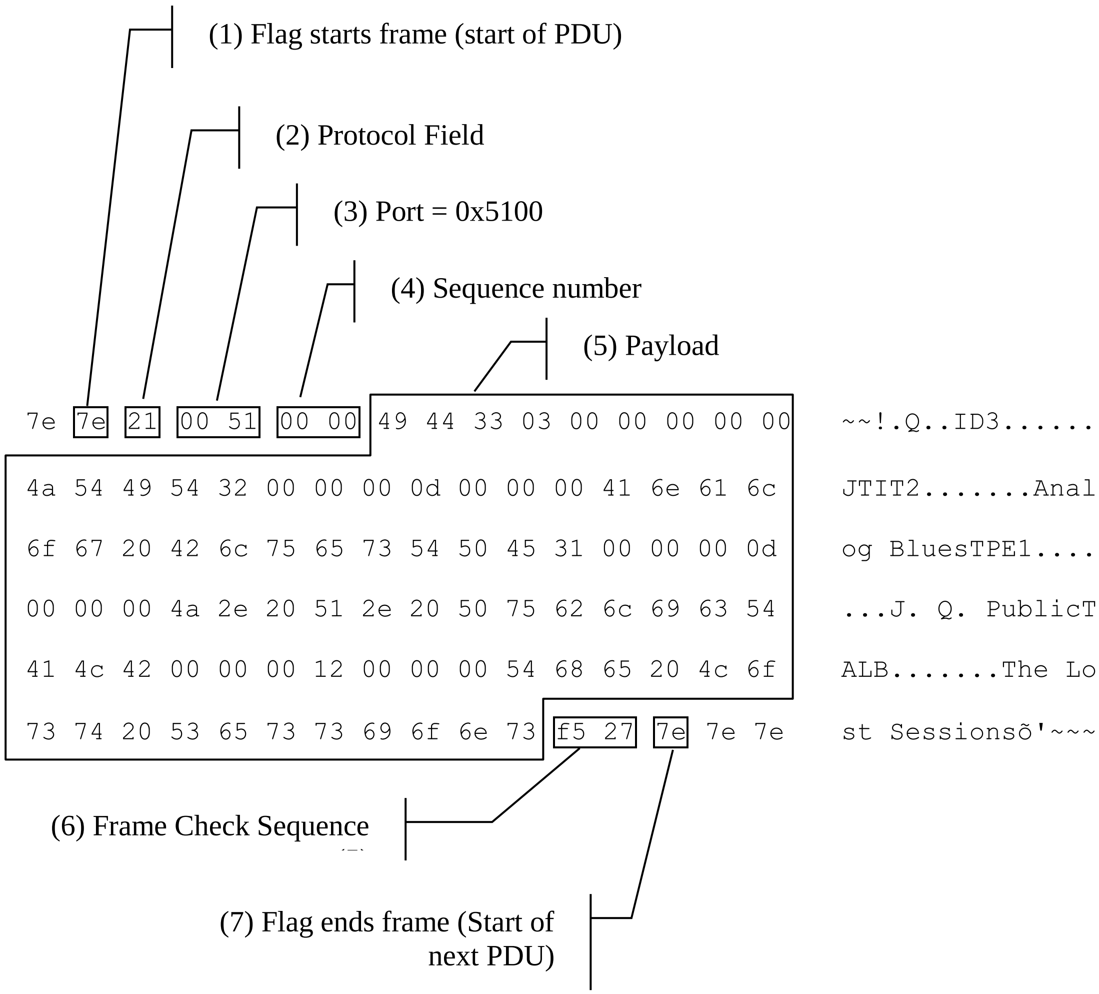

Figure 5-1: PSD PDU Example

A. Appendix

A.1 RFC 1662

```
Network Working Group                                   W. Simpson, Editor
Request for Comments: 1662                                      Daydreamer
STD: 51                                                          July 1994
Obsoletes: 1549
Category: Standards Track
```

PPP in HDLC-like Framing

Status of this Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

Abstract

The Point-to-Point Protocol (PPP) [1] provides a standard method for transporting multi-protocol datagrams over point-to-point links.

This document describes the use of HDLC-like framing for PPP encapsulated packets.

Table of Contents

1. Introduction

This specification provides for framing over both bit-oriented and octet-oriented synchronous links, and asynchronous links with 8 bits of data and no parity. These links MUST be full-duplex, but MAY be either dedicated or circuit-switched.

An escape mechanism is specified to allow control data such as XON/XOFF to be transmitted transparently over the link, and to remove spurious control data which may be injected into the link by intervening hardware and software.

Some protocols expect error free transmission, and either provide error detection only on a conditional basis, or do not provide it at all. PPP uses the HDLC Frame Check Sequence for error detection. This is commonly available in hardware implementations, and a software implementation is provided.

1.1. Specification of Requirements

In this document, several words are used to signify the requirements of the specification. These words are often capitalized.

MUST This word, or the adjective "required", means that the definition is an absolute requirement of the specification.

MUST NOT This phrase means that the definition is an absolute prohibition of the specification.

SHOULD This word, or the adjective "recommended", means that there may exist valid reasons in particular circumstances to ignore this item, but the full implications must be understood and carefully weighed before choosing a different course.

MAY This word, or the adjective "optional", means that this item is one of an allowed set of alternatives. An implementation which does not include this option MUST be prepared to interoperate with another implementation which does include the option.

1.2. Terminology

This document frequently uses the following terms:

datagram The unit of transmission in the network layer (such as IP). A datagram may be encapsulated in one or more packets passed to the data link layer.

frame The unit of transmission at the data link layer. A frame may include a header and/or a trailer, along with some number of units of data.

packet The basic unit of encapsulation, which is passed across the interface between the network layer and the data link layer. A packet is usually mapped to a frame; the exceptions are when data link layer fragmentation is being performed, or when multiple packets are incorporated into a single frame.

peer The other end of the point-to-point link.

silently discard
The implementation discards the packet without further processing. The implementation SHOULD provide the capability of logging the error, including the contents of the silently discarded packet, and SHOULD record the event in a statistics counter.

2. Physical Layer Requirements

PPP is capable of operating across most DTE/DCE interfaces (such as, EIA RS-232-E, EIA RS-422, and CCITT V.35). The only absolute requirement imposed by PPP is the provision of a full-duplex circuit, either dedicated or circuit-switched, which can operate in either an asynchronous (start/stop), bit-synchronous, or octet-synchronous mode, transparent to PPP Data Link Layer frames.

Interface Format

PPP presents an octet interface to the physical layer. There is no provision for sub-octets to be supplied or accepted.

Transmission Rate

PPP does not impose any restrictions regarding transmission rate, other than that of the particular DTE/DCE interface.

Control Signals

PPP does not require the use of control signals, such as Request To Send (RTS), Clear To Send (CTS), Data Carrier Detect (DCD), and Data Terminal Ready (DTR).

When available, using such signals can allow greater functionality and performance. In particular, such signals SHOULD be used to signal the Up and Down events in the LCP Option Negotiation Automaton [1]. When such signals are not available, the implementation MUST signal the Up event to LCP upon initialization, and SHOULD NOT signal the Down event.

Because signalling is not required, the physical layer MAY be decoupled from the data link layer, hiding the transient details of the physical transport. This has implications for mobility in cellular radio networks, and other rapidly switching links.

When moving from cell to cell within the same zone, an implementation MAY choose to treat the entire zone as a single link, even though transmission is switched among several frequencies. The link is considered to be with the central control unit for the zone, rather than the individual cell transceivers. However, the link SHOULD re-establish its configuration whenever the link is switched to a different administration.

Due to the bursty nature of data traffic, some implementations have choosen to disconnect the physical layer during periods of

inactivity, and reconnect when traffic resumes, without informing the data link layer. Robust implementations should avoid using this trick over-zealously, since the price for decreased setup latency is decreased security. Implementations SHOULD signal the Down event whenever "significant time" has elapsed since the link was disconnected. The value for "significant time" is a matter of considerable debate, and is based on the tariffs, call setup times, and security concerns of the installation.

3. The Data Link Layer

PPP uses the principles described in ISO 3309-1979 HDLC frame structure, most recently the fourth edition 3309:1991 [2], which specifies modifications to allow HDLC use in asynchronous environments.

The PPP control procedures use the Control field encodings described in ISO 4335-1979 HDLC elements of procedures, most recently the fourth edition 4335:1991 [4].

This should not be construed to indicate that every feature of the above recommendations are included in PPP. Each feature included is explicitly described in the following sections.

To remain consistent with standard Internet practice, and avoid confusion for people used to reading RFCs, all binary numbers in the following descriptions are in Most Significant Bit to Least Significant Bit order, reading from left to right, unless otherwise indicated. Note that this is contrary to standard ISO and CCITT practice which orders bits as transmitted (network bit order). Keep this in mind when comparing this document with the international standards documents.

3.1. Frame Format

A summary of the PPP HDLC-like frame structure is shown below. This figure does not include bits inserted for synchronization (such as start and stop bits for asynchronous links), nor any bits or octets inserted for transparency. The fields are transmitted from left to right.

```
+----------+----------+----------+
|   Flag   | Address  | Control  |
| 01111110 | 11111111 | 00000011 |
+----------+----------+----------+
+----------+-------------+---------+
| Protocol | Information | Padding |
| 8/16 bits|      *      |    *    |
+----------+-------------+---------+
+----------+----------+-----------------
|   FCS    |   Flag   | Inter-frame Fill
|16/32 bits| 01111110 | or next Address
+----------+----------+-----------------
```

The Protocol, Information and Padding fields are described in the Point-to-Point Protocol Encapsulation [1].

Flag Sequence

Each frame begins and ends with a Flag Sequence, which is the binary sequence 01111110 (hexadecimal 0x7e). All implementations continuously check for this flag, which is used for frame synchronization.

Only one Flag Sequence is required between two frames. Two consecutive Flag Sequences constitute an empty frame, which is silently discarded, and not counted as a FCS error.

Address Field

The Address field is a single octet, which contains the binary sequence 11111111 (hexadecimal 0xff), the All-Stations address. Individual station addresses are not assigned. The All-Stations address MUST always be recognized and received.

The use of other address lengths and values may be defined at a later time, or by prior agreement. Frames with unrecognized Addresses SHOULD be silently discarded.

Control Field

The Control field is a single octet, which contains the binary sequence 00000011 (hexadecimal 0x03), the Unnumbered Information (UI) command with the Poll/Final (P/F) bit set to zero.

The use of other Control field values may be defined at a later time, or by prior agreement. Frames with unrecognized Control field values SHOULD be silently discarded.

Frame Check Sequence (FCS) Field

The Frame Check Sequence field defaults to 16 bits (two octets). The FCS is transmitted least significant octet first, which contains the coefficient of the highest term.

A 32-bit (four octet) FCS is also defined. Its use may be negotiated as described in "PPP LCP Extensions" [5].

The use of other FCS lengths may be defined at a later time, or by prior agreement.

The FCS field is calculated over all bits of the Address, Control, Protocol, Information and Padding fields, not including any start and stop bits (asynchronous) nor any bits (synchronous) or octets (asynchronous or synchronous) inserted for transparency. This also does not include the Flag Sequences nor the FCS field itself.

> When octets are received which are flagged in the Async-Control-Character-Map, they are discarded before calculating the FCS.

For more information on the specification of the FCS, see the Appendices.

The end of the Information and Padding fields is found by locating the closing Flag Sequence and removing the Frame Check Sequence field.

3.2. Modification of the Basic Frame

The Link Control Protocol can negotiate modifications to the standard HDLC-like frame structure. However, modified frames will always be clearly distinguishable from standard frames.

Address-and-Control-Field-Compression

When using the standard HDLC-like framing, the Address and Control fields contain the hexadecimal values 0xff and 0x03 respectively. When other Address or Control field values are in use, Address-and-Control-Field-Compression MUST NOT be negotiated.

On transmission, compressed Address and Control fields are simply omitted.

On reception, the Address and Control fields are decompressed by examining the first two octets. If they contain the values 0xff and 0x03, they are assumed to be the Address and Control fields. If not, it is assumed that the fields were compressed and were not transmitted.

> By definition, the first octet of a two octet Protocol field will never be 0xff (since it is not even). The Protocol field value 0x00ff is not allowed (reserved) to avoid ambiguity when Protocol-Field-Compression is enabled and the first Information field octet is 0x03.

4. Octet-stuffed framing

This chapter summarizes the use of HDLC-like framing with 8-bit asynchronous and octet-synchronous links.

4.1. Flag Sequence

The Flag Sequence indicates the beginning or end of a frame. The octet stream is examined on an octet-by-octet basis for the value 01111110 (hexadecimal 0x7e).

4.2. Transparency

An octet stuffing procedure is used. The Control Escape octet is defined as binary 01111101 (hexadecimal 0x7d), most significant bit first.

As a minimum, sending implementations MUST escape the Flag Sequence and Control Escape octets.

After FCS computation, the transmitter examines the entire frame between the two Flag Sequences. Each Flag Sequence, Control Escape octet, and any octet which is flagged in the sending Async-Control-Character-Map (ACCM), is replaced by a two octet sequence consisting of the Control Escape octet followed by the original octet exclusive-or'd with hexadecimal 0x20.

> This is bit 5 complemented, where the bit positions are numbered 76543210 (the 6th bit as used in ISO numbered 87654321 -- BEWARE when comparing documents).

Receiving implementations MUST correctly process all Control Escape sequences.

On reception, prior to FCS computation, each octet with value less than hexadecimal 0x20 is checked. If it is flagged in the receiving ACCM, it is simply removed (it may have been inserted by intervening data communications equipment). Each Control Escape octet is also removed, and the following octet is exclusive-or'd with hexadecimal 0x20, unless it is the Flag Sequence (which aborts a frame).

A few examples may make this more clear. Escaped data is transmitted on the link as follows:

```
0x7e is encoded as 0x7d, 0x5e.    (Flag Sequence)
0x7d is encoded as 0x7d, 0x5d.    (Control Escape)
0x03 is encoded as 0x7d, 0x23.    (ETX)
```

Some modems with software flow control may intercept outgoing DC1 and DC3 ignoring the 8th (parity) bit. This data would be transmitted on the link as follows:

```
0x11 is encoded as 0x7d, 0x31.    (XON)
0x13 is encoded as 0x7d, 0x33.    (XOFF)
0x91 is encoded as 0x7d, 0xb1.    (XON with parity set)
0x93 is encoded as 0x7d, 0xb3.    (XOFF with parity set)
```

4.3. Invalid Frames

Frames which are too short (less than 4 octets when using the 16-bit FCS), or which end with a Control Escape octet followed immediately by a closing Flag Sequence, or in which octet-framing is violated (by transmitting a "0" stop bit where a "1" bit is expected), are silently discarded, and not counted as a FCS error.

4.4. Time Fill

4.4.1. Octet-synchronous

There is no provision for inter-octet time fill.

The Flag Sequence MUST be transmitted during inter-frame time fill.

4.4.2. Asynchronous

Inter-octet time fill MUST be accomplished by transmitting continuous "1" bits (mark-hold state).

Inter-frame time fill can be viewed as extended inter-octet time fill. Doing so can save one octet for every frame, decreasing delay and increasing bandwidth. This is possible since a Flag Sequence may serve as both a frame end and a frame begin. After having received any frame, an idle receiver will always be in a frame begin state.

Robust transmitters should avoid using this trick over-zealously, since the price for decreased delay is decreased reliability. Noisy links may cause the receiver to receive garbage characters and interpret them as part of an incoming frame. If the transmitter does not send a new opening Flag Sequence before sending the next frame, then that frame will be appended to the noise characters causing an invalid frame (with high reliability).

It is suggested that implementations will achieve the best results by always sending an opening Flag Sequence if the new frame is not back-to-back with the last. Transmitters SHOULD send an open Flag Sequence whenever "appreciable time" has elapsed after the prior closing Flag Sequence. The maximum value for "appreciable time" is likely to be no greater than the typing rate of a slow typist, about 1 second.

4.5. Transmission Considerations

4.5.1. Octet-synchronous

The definition of various encodings and scrambling is the responsibility of the DTE/DCE equipment in use, and is outside the scope of this specification.

4.5.2. Asynchronous

All octets are transmitted least significant bit first, with one start bit, eight bits of data, and one stop bit. There is no provision for seven bit asynchronous links.

5. Bit-stuffed framing

This chapter summarizes the use of HDLC-like framing with bit-synchronous links.

5.1. Flag Sequence

The Flag Sequence indicates the beginning or end of a frame, and is used for frame synchronization. The bit stream is examined on a bit-by-bit basis for the binary sequence 01111110 (hexadecimal 0x7e).

The "shared zero mode" Flag Sequence "011111101111110" SHOULD NOT be used. When not avoidable, such an implementation MUST ensure that the first Flag Sequence detected (the end of the frame) is promptly communicated to the link layer. Use of the shared zero mode hinders interoperability with bit-synchronous to asynchronous and bit-synchronous to octet-synchronous converters.

5.2. Transparency

After FCS computation, the transmitter examines the entire frame between the two Flag Sequences. A "0" bit is inserted after all sequences of five contiguous "1" bits (including the last 5 bits of the FCS) to ensure that a Flag Sequence is not simulated.

On reception, prior to FCS computation, any "0" bit that directly follows five contiguous "1" bits is discarded.

5.3. Invalid Frames

Frames which are too short (less than 4 octets when using the 16-bit FCS), or which end with a sequence of more than six "1" bits, are silently discarded, and not counted as a FCS error.

5.4. Time Fill

There is no provision for inter-octet time fill.

The Flag Sequence SHOULD be transmitted during inter-frame time fill. However, certain types of circuit-switched links require the use of

mark idle (continuous ones), particularly those that calculate accounting based on periods of bit activity. When mark idle is used on a bit-synchronous link, the implementation MUST ensure at least 15 consecutive "1" bits between Flags during the idle period, and that the Flag Sequence is always generated at the beginning of a frame after an idle period.

> This differs from practice in ISO 3309, which allows 7 to 14 bit mark idle.

5.5. Transmission Considerations

All octets are transmitted least significant bit first.

The definition of various encodings and scrambling is the responsibility of the DTE/DCE equipment in use, and is outside the scope of this specification.

While PPP will operate without regard to the underlying representation of the bit stream, lack of standards for transmission will hinder interoperability as surely as lack of data link standards. At speeds of 56 Kbps through 2.0 Mbps, NRZ is currently most widely available, and on that basis is recommended as a default.

When configuration of the encoding is allowed, NRZI is recommended as an alternative, because of its relative immunity to signal inversion configuration errors, and instances when it MAY allow connection without an expensive DSU/CSU. Unfortunately, NRZI encoding exacerbates the missing x1 factor of the 16-bit FCS, so that one error in 2**15 goes undetected (instead of one in 2**16), and triple errors are not detected. Therefore, when NRZI is in use, it is recommended that the 32-bit FCS be negotiated, which includes the x1 factor.

At higher speeds of up to 45 Mbps, some implementors have chosen the ANSI High Speed Synchronous Interface [HSSI]. While this experience is currently limited, implementors are encouraged to cooperate in choosing transmission encoding.

6. Asynchronous to Synchronous Conversion

There may be some use of asynchronous-to-synchronous converters (some built into modems and cellular interfaces), resulting in an asynchronous PPP implementation on one end of a link and a synchronous implementation on the other. It is the responsibility of the converter to do all stuffing conversions during operation.

To enable this functionality, synchronous PPP implementations MUST always respond to the Async-Control-Character-Map Configuration Option with the LCP Configure-Ack. However, acceptance of the Configuration Option does not imply that the synchronous implementation will do any ACCM mapping. Instead, all such octet mapping will be performed by the asynchronous-to-synchronous converter.

7. Additional LCP Configuration Options

The Configuration Option format and basic options are already defined for LCP [1].

Up-to-date values of the LCP Option Type field are specified in the most recent "Assigned Numbers" RFC [10]. This document concerns the following values:

2   Async-Control-Character-Map

7.1. Async-Control-Character-Map (ACCM)

Description

This Configuration Option provides a method to negotiate the use of control character transparency on asynchronous links.

Each end of the asynchronous link maintains two Async-Control-Character-Maps. The receiving ACCM is 32 bits, but the sending ACCM may be up to 256 bits. This results in four distinct ACCMs, two in each direction of the link.

For asynchronous links, the default receiving ACCM is 0xffffffff. The default sending ACCM is 0xffffffff, plus the Control Escape and Flag Sequence characters themselves, plus whatever other outgoing characters are flagged (by prior configuration) as likely to be intercepted.

For other types of links, the default value is 0, since there is no need for mapping.

> The default inclusion of all octets less than hexadecimal 0x20 allows all ASCII control characters [6] excluding DEL (Delete) to be transparently communicated through all known data communications equipment.

The transmitter MAY also send octets with values in the range 0x40 through 0xff (except 0x5e) in Control Escape format. Since these octet values are not negotiable, this does not solve the problem of receivers which cannot handle all non-control characters. Also, since the technique does not affect the 8th bit, this does not solve problems for communications links that can send only 7-bit characters.

> Note that this specification differs in detail from later amendments, such as 3309:1991/Amendment 2 [3]. However, such "extended transparency" is applied only by "prior agreement". Use of the transparency methods in this specification constitute a prior agreement with respect to PPP.
>
> For compatibility with 3309:1991/Amendment 2, the transmitter MAY escape DEL and ACCM equivalents with the 8th (most significant) bit set. No change is required in the receiving algorithm.
>
> Following ACCM negotiation, the transmitter SHOULD cease escaping DEL.

However, it is rarely necessary to map all control characters, and often it is unnecessary to map any control characters. The Configuration Option is used to inform the peer which control characters MUST remain mapped when the peer sends them.

The peer MAY still send any other octets in mapped format, if it is necessary because of constraints known to the peer. The peer SHOULD Configure-Nak with the logical union of the sets of mapped octets, so that when such octets are spuriously introduced they can be ignored on receipt.

A summary of the Async-Control-Character-Map Configuration Option format is shown below. The fields are transmitted from left to right.

```
 0                   1                   2                   3
```

```
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|     Type      |    Length     |               ACCM
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
          ACCM (cont)           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Type

2

Length

6

ACCM

The ACCM field is four octets, and indicates the set of control characters to be mapped. The map is sent most significant octet first.

Each numbered bit corresponds to the octet of the same value. If the bit is cleared to zero, then that octet need not be mapped. If the bit is set to one, then that octet MUST remain mapped. For example, if bit 19 is set to zero, then the ASCII control character 19 (DC3, Control-S) MAY be sent in the clear.

Note: The least significant bit of the least significant octet (the final octet transmitted) is numbered bit 0, and would map to the ASCII control character NUL.

A. Recommended LCP Options

The following Configurations Options are recommended:

High Speed links

Magic Number
Link Quality Monitoring
No Address and Control Field Compression
No Protocol Field Compression

Low Speed or Asynchronous links

Async Control Character Map
Magic Number
Address and Control Field Compression
Protocol Field Compression

B. Automatic Recognition of PPP Frames

It is sometimes desirable to detect PPP frames, for example during a login sequence. The following octet sequences all begin valid PPP LCP frames:

```
7e ff 03 c0 21
7e ff 7d 23 c0 21
```

```
7e 7d df 7d 23 c0 21
```

Note that the first two forms are not a valid username for Unix. However, only the third form generates a correctly checksummed PPP frame, whenever 03 and ff are taken as the control characters ETX and DEL without regard to parity (they are correct for an even parity link) and discarded.

Many implementations deal with this by putting the interface into packet mode when one of the above username patterns are detected during login, without examining the initial PPP checksum. The initial incoming PPP frame is discarded, but a Configure-Request is sent immediately.

C. Fast Frame Check Sequence (FCS) Implementation

The FCS was originally designed with hardware implementations in mind. A serial bit stream is transmitted on the wire, the FCS is calculated over the serial data as it goes out, and the complement of the resulting FCS is appended to the serial stream, followed by the Flag Sequence.

The receiver has no way of determining that it has finished calculating the received FCS until it detects the Flag Sequence. Therefore, the FCS was designed so that a particular pattern results when the FCS operation passes over the complemented FCS. A good frame is indicated by this "good FCS" value.

C.1. FCS table generator

The following code creates the lookup table used to calculate the FCS-16.

```
/*
 * Generate a FCS-16 table.
 *
 * Drew D. Perkins at Carnegie Mellon University.
 *
 * Code liberally borrowed from Mohsen Banan and D. Hugh Redelmeier.
 */

/*
 * The FCS-16 generator polynomial: x**0 + x**5 + x**12 + x**16.
 */
#define P       0x8408

main()
{
    register unsigned int b, v;
    register int i;

    printf("typedef unsigned short u16;\n");
    printf("static u16 fcstab[256] = {");
    for (b = 0; ; ) {
        if (b % 8 == 0)
            printf("\n");
```

```
        v = b;
        for (i = 8; i--; )

            v = v &amp; 1 ? (v &gt;&gt; 1) ^ P : v &gt;&gt; 1;

        printf("\t0x%04x", v &amp; 0xFFFF);
        if (++b == 256)
            break;
        printf(",");
    }
    printf("\n};\n");
}
```

C.2. 16-bit FCS Computation Method

The following code provides a table lookup computation for calculating the Frame Check Sequence as data arrives at the interface. This implementation is based on [7], [8], and [9].

```
/*
 * u16 represents an unsigned 16-bit number.  Adjust the typedef for
 * your hardware.
 */
typedef unsigned short u16;

/*
 * FCS lookup table as calculated by the table generator.
 */
static u16 fcstab[256] = {
   0x0000, 0x1189, 0x2312, 0x329b, 0x4624, 0x57ad, 0x6536, 0x74bf,
   0x8c48, 0x9dc1, 0xaf5a, 0xbed3, 0xca6c, 0xdbe5, 0xe97e, 0xf8f7,
   0x1081, 0x0108, 0x3393, 0x221a, 0x56a5, 0x472c, 0x75b7, 0x643e,
   0x9cc9, 0x8d40, 0xbfdb, 0xae52, 0xdaed, 0xcb64, 0xf9ff, 0xe876,
   0x2102, 0x308b, 0x0210, 0x1399, 0x6726, 0x76af, 0x4434, 0x55bd,
   0xad4a, 0xbcc3, 0x8e58, 0x9fd1, 0xeb6e, 0xfae7, 0xc87c, 0xd9f5,
   0x3183, 0x200a, 0x1291, 0x0318, 0x77a7, 0x662e, 0x54b5, 0x453c,
   0xbdcb, 0xac42, 0x9ed9, 0x8f50, 0xfbef, 0xea66, 0xd8fd, 0xc974,
   0x4204, 0x538d, 0x6116, 0x709f, 0x0420, 0x15a9, 0x2732, 0x36bb,
   0xce4c, 0xdfc5, 0xed5e, 0xfcd7, 0x8868, 0x99e1, 0xab7a, 0xbaf3,
   0x5285, 0x430c, 0x7197, 0x601e, 0x14a1, 0x0528, 0x37b3, 0x263a,
   0xdecd, 0xcf44, 0xfddf, 0xec56, 0x98e9, 0x8960, 0xbbfb, 0xaa72,
   0x6306, 0x728f, 0x4014, 0x519d, 0x2522, 0x34ab, 0x0630, 0x17b9,
   0xef4e, 0xfec7, 0xcc5c, 0xddd5, 0xa96a, 0xb8e3, 0x8a78, 0x9bf1,
   0x7387, 0x620e, 0x5095, 0x411c, 0x35a3, 0x242a, 0x16b1, 0x0738,
   0xffcf, 0xee46, 0xdcdd, 0xcd54, 0xb9eb, 0xa862, 0x9af9, 0x8b70,
   0x8408, 0x9581, 0xa71a, 0xb693, 0xc22c, 0xd3a5, 0xe13e, 0xf0b7,
   0x0840, 0x19c9, 0x2b52, 0x3adb, 0x4e64, 0x5fed, 0x6d76, 0x7cff,
   0x9489, 0x8500, 0xb79b, 0xa612, 0xd2ad, 0xc324, 0xf1bf, 0xe036,
   0x18c1, 0x0948, 0x3bd3, 0x2a5a, 0x5ee5, 0x4f6c, 0x7df7, 0x6c7e,

   0xa50a, 0xb483, 0x8618, 0x9791, 0xe32e, 0xf2a7, 0xc03c, 0xd1b5,
   0x2942, 0x38cb, 0x0a50, 0x1bd9, 0x6f66, 0x7eef, 0x4c74, 0x5dfd,
   0xb58b, 0xa402, 0x9699, 0x8710, 0xf3af, 0xe226, 0xd0bd, 0xc134,
   0x39c3, 0x284a, 0x1ad1, 0x0b58, 0x7fe7, 0x6e6e, 0x5cf5, 0x4d7c,
   0xc60c, 0xd785, 0xe51e, 0xf497, 0x8028, 0x91a1, 0xa33a, 0xb2b3,
   0x4a44, 0x5bcd, 0x6956, 0x78df, 0x0c60, 0x1de9, 0x2f72, 0x3efb,
   0xd68d, 0xc704, 0xf59f, 0xe416, 0x90a9, 0x8120, 0xb3bb, 0xa232,
```

```
      0x5ac5, 0x4b4c, 0x79d7, 0x685e, 0x1ce1, 0x0d68, 0x3ff3, 0x2e7a,
      0xe70e, 0xf687, 0xc41c, 0xd595, 0xa12a, 0xb0a3, 0x8238, 0x93b1,
      0x6b46, 0x7acf, 0x4854, 0x59dd, 0x2d62, 0x3ceb, 0x0e70, 0x1ff9,
      0xf78f, 0xe606, 0xd49d, 0xc514, 0xb1ab, 0xa022, 0x92b9, 0x8330,
      0x7bc7, 0x6a4e, 0x58d5, 0x495c, 0x3de3, 0x2c6a, 0x1ef1, 0x0f78
   };

   #define PPPINITFCS16    0xffff  /* Initial FCS value */
   #define PPPGOODFCS16    0xf0b8  /* Good final FCS value */

   /*
    * Calculate a new fcs given the current fcs and the new data.
    */
   u16 pppfcs16(fcs, cp, len)
       register u16 fcs;
       register unsigned char *cp;
       register int len;
   {
       ASSERT(sizeof (u16) == 2);
       ASSERT(((u16) -1) &gt; 0);
       while (len--)
           fcs = (fcs &gt;&gt; 8) ^ fcstab[(fcs ^ *cp++) &amp; 0xff];

       return (fcs);
   }

   /*
    * How to use the fcs
    */
   tryfcs16(cp, len)
       register unsigned char *cp;
       register int len;
   {
       u16 trialfcs;

       /* add on output */
       trialfcs = pppfcs16( PPPINITFCS16, cp, len );
       trialfcs ^= 0xffff;                 /* complement */
       cp[len] = (trialfcs &amp; 0x00ff);      /* least significant byte
first */
       cp[len+1] = ((trialfcs &gt;&gt; 8) &amp; 0x00ff);

       /* check on input */
       trialfcs = pppfcs16( PPPINITFCS16, cp, len + 2 );
       if ( trialfcs == PPPGOODFCS16 )
           printf("Good FCS\n");
   }
```

C.3. 32-bit FCS Computation Method

The following code provides a table lookup computation for calculating the 32-bit Frame Check Sequence as data arrives at the interface.

```
   /*
    * The FCS-32 generator polynomial: x**0 + x**1 + x**2 + x**4 + x**5
    *                      + x**7 + x**8 + x**10 + x**11 + x**12 + x**16
```

```
 *                      + x**22 + x**23 + x**26 + x**32.
 */

/*
 * u32 represents an unsigned 32-bit number.  Adjust the typedef for
 * your hardware.
 */
typedef unsigned long u32;

static u32 fcstab_32[256] =
   {
   0x00000000, 0x77073096, 0xee0e612c, 0x990951ba,
   0x076dc419, 0x706af48f, 0xe963a535, 0x9e6495a3,
   0x0edb8832, 0x79dcb8a4, 0xe0d5e91e, 0x97d2d988,
   0x09b64c2b, 0x7eb17cbd, 0xe7b82d07, 0x90bf1d91,
   0x1db71064, 0x6ab020f2, 0xf3b97148, 0x84be41de,
   0x1adad47d, 0x6ddde4eb, 0xf4d4b551, 0x83d385c7,
   0x136c9856, 0x646ba8c0, 0xfd62f97a, 0x8a65c9ec,
   0x14015c4f, 0x63066cd9, 0xfa0f3d63, 0x8d080df5,
   0x3b6e20c8, 0x4c69105e, 0xd56041e4, 0xa2677172,
   0x3c03e4d1, 0x4b04d447, 0xd20d85fd, 0xa50ab56b,
   0x35b5a8fa, 0x42b2986c, 0xdbbbc9d6, 0xacbcf940,
   0x32d86ce3, 0x45df5c75, 0xdcd60dcf, 0xabd13d59,
   0x26d930ac, 0x51de003a, 0xc8d75180, 0xbfd06116,
   0x21b4f4b5, 0x56b3c423, 0xcfba9599, 0xb8bda50f,
   0x2802b89e, 0x5f058808, 0xc60cd9b2, 0xb10be924,
   0x2f6f7c87, 0x58684c11, 0xc1611dab, 0xb6662d3d,
   0x76dc4190, 0x01db7106, 0x98d220bc, 0xefd5102a,
   0x71b18589, 0x06b6b51f, 0x9fbfe4a5, 0xe8b8d433,
   0x7807c9a2, 0x0f00f934, 0x9609a88e, 0xe10e9818,
   0x7f6a0dbb, 0x086d3d2d, 0x91646c97, 0xe6635c01,

   0x6b6b51f4, 0x1c6c6162, 0x856530d8, 0xf262004e,
   0x6c0695ed, 0x1b01a57b, 0x8208f4c1, 0xf50fc457,
   0x65b0d9c6, 0x12b7e950, 0x8bbeb8ea, 0xfcb9887c,
   0x62dd1ddf, 0x15da2d49, 0x8cd37cf3, 0xfbd44c65,
   0x4db26158, 0x3ab551ce, 0xa3bc0074, 0xd4bb30e2,
   0x4adfa541, 0x3dd895d7, 0xa4d1c46d, 0xd3d6f4fb,
   0x4369e96a, 0x346ed9fc, 0xad678846, 0xda60b8d0,
   0x44042d73, 0x33031de5, 0xaa0a4c5f, 0xdd0d7cc9,
   0x5005713c, 0x270241aa, 0xbe0b1010, 0xc90c2086,
   0x5768b525, 0x206f85b3, 0xb966d409, 0xce61e49f,
   0x5edef90e, 0x29d9c998, 0xb0d09822, 0xc7d7a8b4,
   0x59b33d17, 0x2eb40d81, 0xb7bd5c3b, 0xc0ba6cad,
   0xedb88320, 0x9abfb3b6, 0x03b6e20c, 0x74b1d29a,
   0xead54739, 0x9dd277af, 0x04db2615, 0x73dc1683,
   0xe3630b12, 0x94643b84, 0x0d6d6a3e, 0x7a6a5aa8,
   0xe40ecf0b, 0x9309ff9d, 0x0a00ae27, 0x7d079eb1,
   0xf00f9344, 0x8708a3d2, 0x1e01f268, 0x6906c2fe,
   0xf762575d, 0x806567cb, 0x196c3671, 0x6e6b06e7,
   0xfed41b76, 0x89d32be0, 0x10da7a5a, 0x67dd4acc,
   0xf9b9df6f, 0x8ebeeff9, 0x17b7be43, 0x60b08ed5,
   0xd6d6a3e8, 0xa1d1937e, 0x38d8c2c4, 0x4fdff252,
   0xd1bb67f1, 0xa6bc5767, 0x3fb506dd, 0x48b2364b,
   0xd80d2bda, 0xaf0a1b4c, 0x36034af6, 0x41047a60,
   0xdf60efc3, 0xa867df55, 0x316e8eef, 0x4669be79,
   0xcb61b38c, 0xbc66831a, 0x256fd2a0, 0x5268e236,
```

```
        0xcc0c7795, 0xbb0b4703, 0x220216b9, 0x5505262f,
        0xc5ba3bbe, 0xb2bd0b28, 0x2bb45a92, 0x5cb36a04,
        0xc2d7ffa7, 0xb5d0cf31, 0x2cd99e8b, 0x5bdeae1d,
        0x9b64c2b0, 0xec63f226, 0x756aa39c, 0x026d930a,
        0x9c0906a9, 0xeb0e363f, 0x72076785, 0x05005713,
        0x95bf4a82, 0xe2b87a14, 0x7bb12bae, 0x0cb61b38,
        0x92d28e9b, 0xe5d5be0d, 0x7cdcefb7, 0x0bdbdf21,
        0x86d3d2d4, 0xf1d4e242, 0x68ddb3f8, 0x1fda836e,
        0x81be16cd, 0xf6b9265b, 0x6fb077e1, 0x18b74777,
        0x88085ae6, 0xff0f6a70, 0x66063bca, 0x11010b5c,
        0x8f659eff, 0xf862ae69, 0x616bffd3, 0x166ccf45,
        0xa00ae278, 0xd70dd2ee, 0x4e048354, 0x3903b3c2,
        0xa7672661, 0xd06016f7, 0x4969474d, 0x3e6e77db,
        0xaed16a4a, 0xd9d65adc, 0x40df0b66, 0x37d83bf0,
        0xa9bcae53, 0xdebb9ec5, 0x47b2cf7f, 0x30b5ffe9,
        0xbdbdf21c, 0xcabac28a, 0x53b39330, 0x24b4a3a6,
        0xbad03605, 0xcdd70693, 0x54de5729, 0x23d967bf,
        0xb3667a2e, 0xc4614ab8, 0x5d681b02, 0x2a6f2b94,
        0xb40bbe37, 0xc30c8ea1, 0x5a05df1b, 0x2d02ef8d
        };

   #define PPPINITFCS32  0xffffffff   /* Initial FCS value */
   #define PPPGOODFCS32  0xdebb20e3   /* Good final FCS value */

   /*
    * Calculate a new FCS given the current FCS and the new data.
    */
   u32 pppfcs32(fcs, cp, len)
       register u32 fcs;
       register unsigned char *cp;
       register int len;
       {
       ASSERT(sizeof (u32) == 4);
       ASSERT(((u32) -1) &gt; 0);
       while (len--)
           fcs = (((fcs) &gt;&gt; 8) ^ fcstab_32[((fcs) ^ (*cp++)) &amp;
0xff]);

       return (fcs);
       }

   /*
    * How to use the fcs
    */
   tryfcs32(cp, len)
       register unsigned char *cp;
       register int len;
   {
       u32 trialfcs;

       /* add on output */
       trialfcs = pppfcs32( PPPINITFCS32, cp, len );
       trialfcs ^= 0xffffffff;             /* complement */
       cp[len] = (trialfcs &amp; 0x00ff);      /* least significant byte
first */
       cp[len+1] = ((trialfcs &gt;&gt;= 8) &amp; 0x00ff);
       cp[len+2] = ((trialfcs &gt;&gt;= 8) &amp; 0x00ff);
```

```
        cp[len+3] = ((trialfcs &gt;&gt; 8) &amp; 0x00ff);

        /* check on input */
        trialfcs = pppfcs32( PPPINITFCS32, cp, len + 4 );
        if ( trialfcs == PPPGOODFCS32 )
            printf("Good FCS\n");
    }
```

Security Considerations

As noted in the Physical Layer Requirements section, the link layer might not be informed when the connected state of the physical layer has changed. This results in possible security lapses due to over-reliance on the integrity and security of switching systems and administrations. An insertion attack might be undetected. An attacker which is able to spoof the same calling identity might be able to avoid link authentication.

References

[1] Simpson, W., Editor, "The Point-to-Point Protocol (PPP)", STD 50, <A HREF="http://www.faqs.org/rfcs/rfc1661.html">RFC 1661</A>, Daydreamer, July 1994.

[2] ISO/IEC 3309:1991(E), "Information Technology - Telecommunications and information exchange between systems - High-level data link control (HDLC) procedures - Frame structure", International Organization For Standardization, Fourth edition 1991-06-01.

[3] ISO/IEC 3309:1991/Amd.2:1992(E), "Information Technology - Telecommunications and information exchange between systems - High-level data link control (HDLC) procedures - Frame structure - Amendment 2: Extended transparency options for start/stop transmission", International Organization For Standardization, 1992-01-15.

[4] ISO/IEC 4335:1991(E), "Information Technology - Telecommunications and information exchange between systems - High-level data link control (HDLC) procedures - Elements of procedures", International Organization For Standardization, Fourth edition 1991-09-15.

[5] Simpson, W., Editor, "PPP LCP Extensions", <A HREF="http://www.faqs.org/rfcs/rfc1570.html">RFC 1570</A>, Daydreamer, January 1994.

[6] ANSI X3.4-1977, "American National Standard Code for Information Interchange", American National Standards Institute, 1977.

[7] Perez, "Byte-wise CRC Calculations", IEEE Micro, June 1983.

[8] Morse, G., "Calculating CRC's by Bits and Bytes", Byte, September 1986.

[9] LeVan, J., "A Fast CRC", Byte, November 1987.

[10] Reynolds, J., and J. Postel, "Assigned Numbers", STD 2, RFC 1340, USC/Information Sciences Institute, July 1992.

Acknowledgements

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HD Radio™ Air Interface Design Description Advanced Application Services Transport

Rev. H

December 14, 2016

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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 Advanced Application Services Transport (AAT). It describes the packet encapsulation of the fixed and opportunistic data and the generation of AAS PDUs. The HD Radio Link Subsystem (HD RLS) provides the packet transport mechanism for the Advanced Application Services Transport (AAT). The HD RLS also provides the packet encapsulation process for the Program Service Data (PSD) and the generation of PSD PDUs which is described in Reference [10]. Specific hardware and software implementation is not described. See References [1] to [24] for more details.

2 Referenced Documents

STATEMENT
Each referenced document that is mentioned in this document shall be listed in the following iBiquity document:

- Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard Document Number: SY_REF_2690s

3 Abbreviations and Conventions

3.1 Abbreviations and Acronyms

AAS	Advanced Application Services
AAT	Advanced Application Services Transport
AM	Amplitude Modulation
BBM	Block Boundary Marker
CRC	Cyclic Redundancy Check
CCC	Configuration Control Channel
DDL	Data Delimiter
DTPF	Data Transport Packet Format
ECK	Encryption Control Key
FCC	Federal Communications Commission
FCS	Frame Check Sequence
FEC	Forward Error Correction
FM	Frequency Modulation
HDC	HD Codec
HDLC	High-Level Data Link Control
HD RLS	HD Radio Link Subsystem
IBOC	In Band On Channel
IETF	Internet Engineering Task Force
IP	Internet Protocol
LCP	Link Control Protocol
LSB	Least Significant Bit
L1	Layer 1
L2	Layer 2
MF	Medium Frequency
MPS	Main Program Service
SPS	Supplemental Program Service
MPSD	Main Program Service Data
SPSD	Supplemental Program Service Data
OFDM	Orthogonal Frequency Division Multiplexing
PDU	Protocol Data Unit
PPP	Point to Point Protocol
PSD	Program Service Data
RF	Radio Frequency
RFC	Request For Comment
RS	Reed Solomon
UINT	unsigned integer
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.
- Bit 0 of a byte or word is considered the least significant bit.
- In representations of binary numbers, the least significant bit is on the right.
- 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.

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	$\underline{\underline{u}}[i][j]$, $\underline{\underline{V}}[i][j]$
$\begin{bmatrix} \quad & \quad \\ \quad & \quad \end{bmatrix}$	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]

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	$\log_{10}(100) = 2$

4 Overview

4.1 Introduction

The Advanced Applications Services Transport (AAT) is used in the transport of fixed and opportunistic data in the HD Radio system. Figure 4-1 shows the interface of the AAT with the rest of the system. Various Advanced Application Services (AAS) use the Service Interfaces to interact with the HD Radio system. During broadcast, the AAT receives AAS Data from the Service Interfaces and then encodes and encapsulates this data to generate AAS PDUs. The AAS PDUs are then sent to Layer 2 for further processing. The AAS PDUs are sent over different bearer channels which carry fixed data or opportunistic data packets. The opportunistic data bandwidth depends on the audio content transmitted. See Subsection 4.2.1 for details on the bearer channels. Thus, the AAT receives the opportunistic bandwidth status from the Audio Transport allowing the inclusion of opportunistic data.

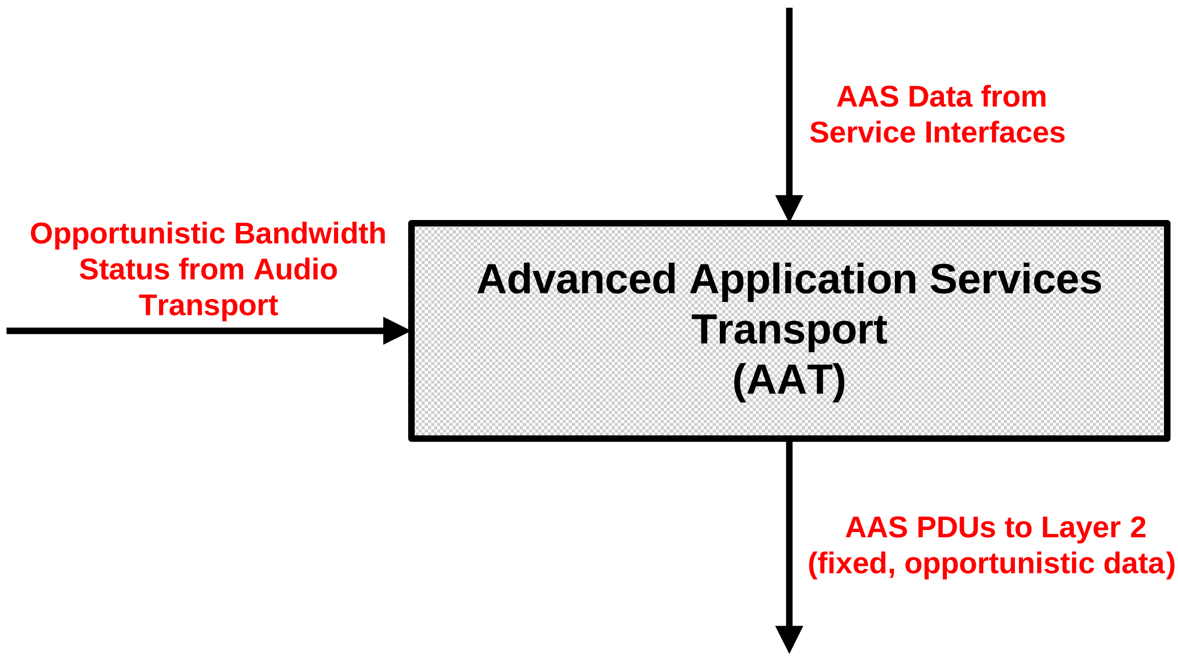

Figure 4-1: AAS Transport Interface

4.2 Packet Transport Mechanism

The packet transport capability is provided by HD RLS which is iBiquity's implementation of the packet transport mechanism for the AAT. It performs the framing and encapsulation of the data packets and generates the AAS PDUs consisting of the fixed and opportunistic data. In addition, the HD RLS also serves as the packet transport mechanism for the PSD Transport and the generation of PSD PDUs which are interleaved along with the audio content ([10]). Figure 4-2 shows the HD RLS packet transport mechanism and the processing of various Data Services which are eventually output to Layer 2 on the different bearer channels. The output of Layer 2 is carried by the respective Layer 1 logical channels (References [1] and [2]) to the Waveform/Transmission layer.

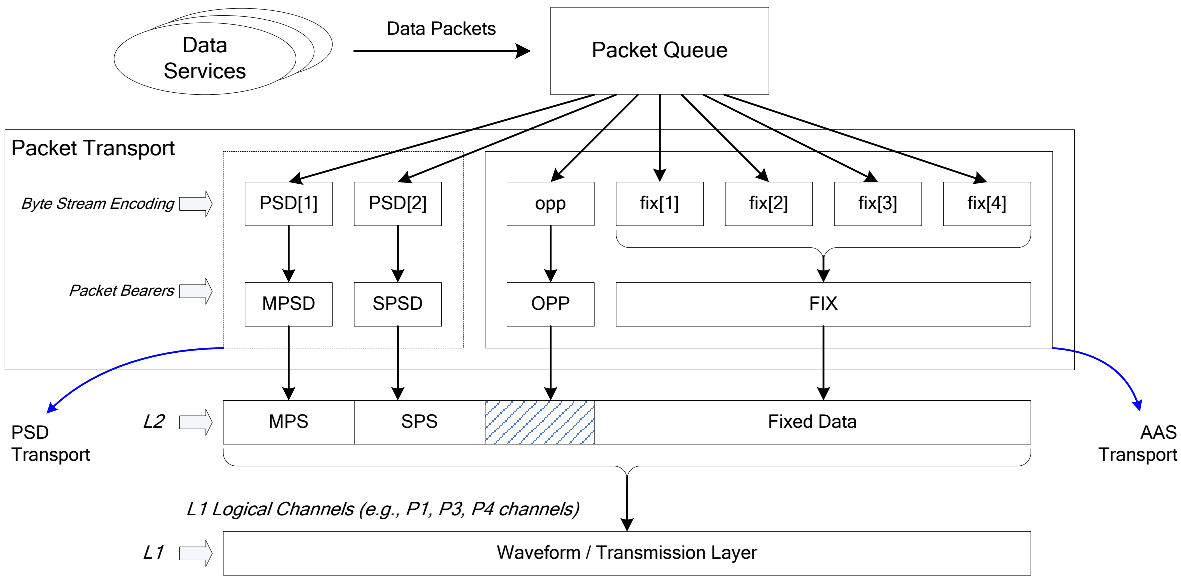

Figure 4-2: HD RLS Packet Transport Mechanism

At the transmit side, data packets from various Data Services are queued for transmission. Packets to be transmitted are encoded in streams that are transmitted over one or more bearer channels which are then packed into a Layer 2 PDU for broadcast through the HD Radio system (See Subsection 7.1). The actual transmission of a packet may be affected by numerous factors, including bandwidth allocation for the bearer channel and packet length. Therefore, packets may be transmitted over multiple L1 frames.

4.2.1 Bearer Channels

The HD Radio system provides multiple channels for carrying data streams (References [1] and [2]). These channels are categorized based on the type of data they carry and are referred to as bearer channels. Each of these bearer channels is used to transport data packets in one or more encoded byte streams.

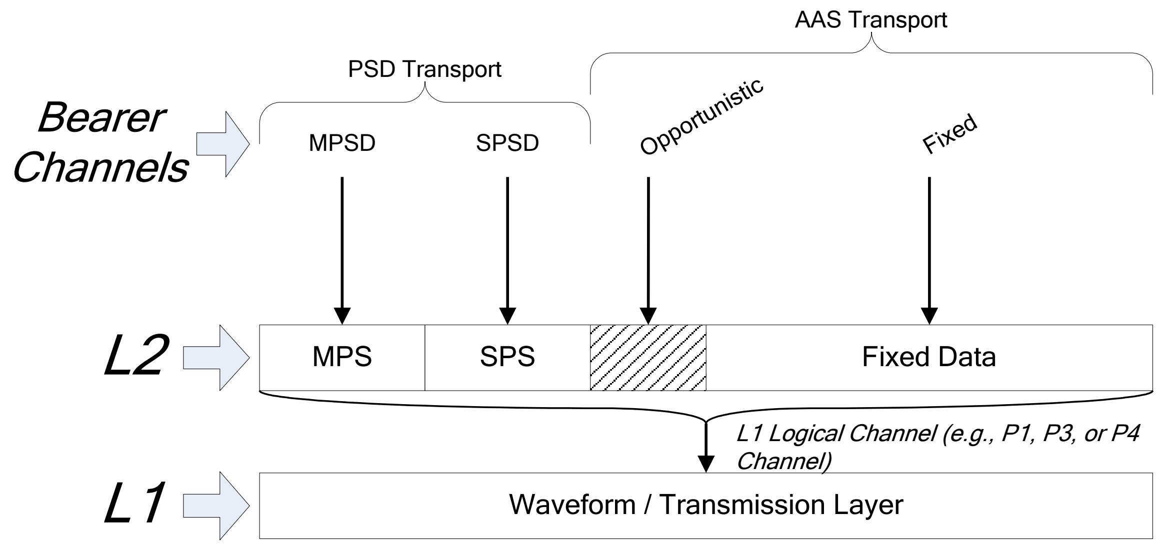

Figure 4-3: Bearer Channels

The different bearer channels are as follows:

- **PSD** – A PSD bearer channel is created from bytes allocated within the Audio Transport frames that carry digital audio for Main Program Services (MPS) and Supplemental Program Services (SPS). The Audio Transport obtains the encapsulated PSD byte streams, if present, and multiplexes them with the encoded audio packets. The PSD byte streams can be either associated with the Main Program Service Data (MPSD) or with the Supplemental Program Service Data (SPSD). The HD RLS mechanism within the PSD Transport encapsulates the PSD as PSD PDUs which are then multiplexed along with the audio program. The Audio Transport provides the mechanisms for inserting PSD at the transmitter and extracting it at the receiver.
- **Opportunistic** – If the instantaneous audio content requires less than its regularly allocated portion of the L2 PDU, the instantaneously available capacity is used to create an opportunistic bearer channel. The opportunistic bytes, if they exist, are always located in the L2 PDU before the fixed data. If there is no fixed data allocation, opportunistic bytes are located at the end of the L2 PDU.
- **Fixed** – A fixed data bearer channel uses a dedicated portion of the L2 PDU which has been allocated for data services. The fixed data is always located at the end of the L2 PDU and maintains a constant size over long periods (that is, many PDUs).

The AAS Transport uses the HD RLS mechanism to encapsulate the fixed and opportunistic data (respectively, from these bearer channels) as AAS PDUs which are sent to Layer 2.

Section 7 provides detailed descriptions of how data is sent in each type of bearer channel.

4.2.2 Packet Structure

The data packets in the HD Radio system are structured as variable length datagrams, as briefly shown in Figure 4-4. Section 5 contains a detailed description of data packet structure.

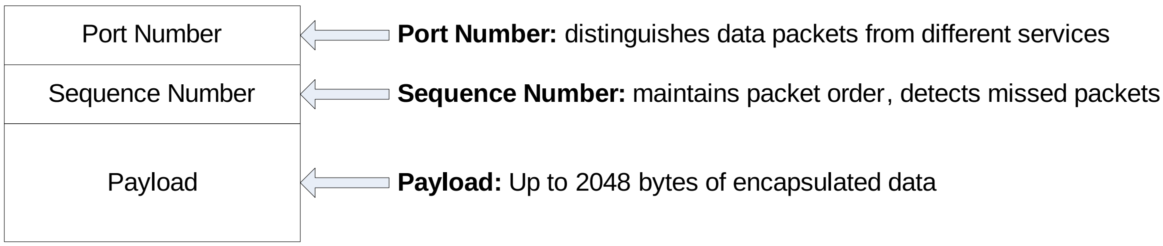

Figure 4-4: Packet Structure

4.2.3 Byte Stream Encoding

Packets are encoded in a serial byte stream before they are carried on the bearer channels. To improve reliability, the encoded byte stream may be protected with an adjustable level of Forward Error Correction (FEC) that consists of a block error code, interleaving, and data blocking, The FEC level can be customized for each bearer channel.

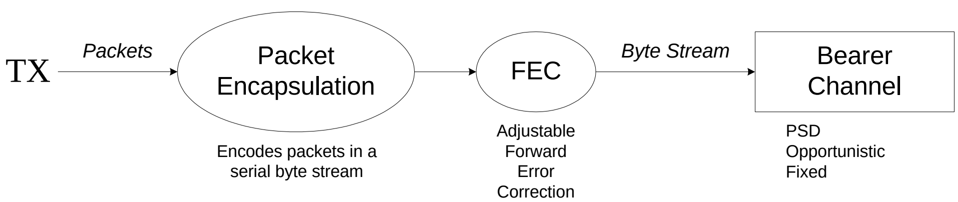

Figure 4-5: Byte Stream Encoding

Section 6 contains a detailed description of packet encapsulation and FEC used to encode packets in byte streams.

5 Data Packet Structure

The AAS Data packets are structured by HD RLS to allow applications at the transmit side to send data to applications at the receiver over the AAT. The data packet fields are listed in Table 5-1.

Table 5-1: Data Packet Format

Field	Size in Bytes and Format*
PORT	2 (little-endian format)
SEQ	2 (little-endian format)
Payload[ ]	1 to 2048 (byte format)

* Payload is limited to a maximum size of 2048 bytes after the addition of any escape bytes as defined in Subsection 6.2.5. The SEQ field is nominally 2 bytes in length but may be 3 or 4 bytes in length if one or both bytes need to be escaped, respectively. The PORT field is always 2 bytes in length because port numbers that require escaping are not allowed in the system.

5.1 Port Number

Port numbers in HD RLS are used to identify data packets with respect to the data services supported by their content. Port numbers indicate to the receiver the application to which a received packet should be directed. The format is little-endian.

While certain port numbers are always associated with specific use, other port numbers may be allocated by HD Radio AAS for specific allocation. The mapping of port numbers to applications is not defined by HD RLS. For example, port number 0x5100 is used for Main Program Service Data (MPSD); ports 0x0000 through 0x03FF are reserved for use by the HD Radio system and are not available to applications. Table 5-2 lists the assignment of the port numbers.

NRSC Supplemental Information provides the most up-to-date information regarding the port number assignments: http://hdradio.com/broadcasters/us-regulatory/nrsc-supplemental-information.

Table 5-2: Port Number Assignment

Port Number	Status
0x0000 to 0x0400*	Reserved for System Use
0x0401 to 0x50FF*	Available for Specific Applications
0x5100	MPSD
0x5101 to 0x5200*	Reserved for System Use
0x5201 to 0x5207	SPSD
0x5208 to 0x52FF*	Reserved for System Use
0x5300 to 0x7CFF*	Reserved for Future Applications
0x7D00 to 0x7EFF	Invalid – Shall not be used
0x7F00 to 0xFEFF*	Reserved for Future Applications
0xFF00 to 0xFFFF*	Reserved for System Use
0x XX7D, 0x XX7E*	XX = any byte value Invalid – Shall not be used, see Note

* Note: Any Port Number containing a byte value of 0x7D or 0x7E is invalid and shall not be used. For example, port number 0x537D is invalid but 0x537F is OK

5.2 Sequence Number

At the transmitter, a packet sequence number is maintained for each packet sent to a given port. It is incremented (by 1) for every new packet for that port. The sequence number is incremented independently per port. This allows for packet order to be verified at the receiver and for lost packets to be detected through missing sequence numbers. The sequence number is in little-endian format.

5.3 Packet Payload

The packet payload can be of any size up to 2048 encapsulated (i.e., including escape mechanism for specific characters – see Section 6.2.5) bytes in length.

6 Byte Stream Encoding

6.1 Overview

Each data bearer channel in the HD Radio system transports a stream of bytes. The allocated number of bytes in each L1 frame is constant (fixed data) or variable (PSD and opportunistic data). The aggregate number of bytes used in each L1 frame may vary.

To send data over these channels, packets are encoded in a continuous byte stream. Successful packet delivery relies on the bearer channels to deliver the bytes in the same order that they were transmitted. The portion of these encoded byte streams, which a bearer channel transports in an L1 frame, may contain one or more packets, a portion of a single packet, portions of multiple packets, or fragments of multiple packets (when FEC is used).

The general structure of an encoded byte stream is shown in Figure 6-1.

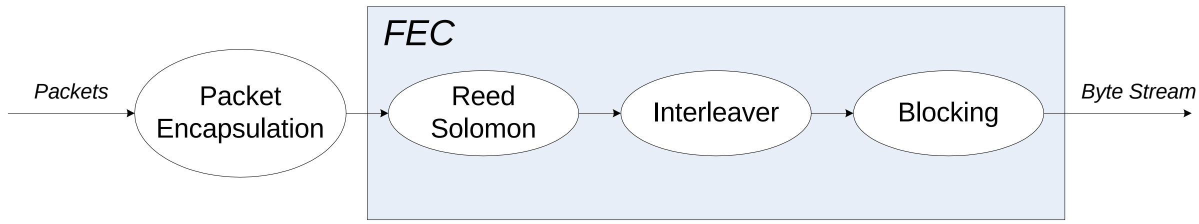

Figure 6-1: Byte Stream Encoding

Packet Encapsulation encodes AAS Data packets as a serial byte stream with embedded error detection.

Forward Error Correction (FEC) may be applied to the encoded packet stream to control packet loss and errors using the following methods:

- Reed Solomon block coding for error correction
- Byte Interleaving to protect against error bursts
- Block synchronization mechanism

6.2 Packet Encapsulation – PDU Generation

The packet encapsulation used by the HD RLS mechanism follows the HDLC-like framing employed by the Point-to-Point Protocol (PPP) as standardized by the IETF in RFC 1662 (Reference [24]). The following sections describe how the HDLC-like framing of PPP has been adapted for the HD Radio system.

The HDLC-like framing allows encapsulation of a packet within a byte stream, referred to as AAS PDU, that may be broadcast in segments of arbitrary size (for example, in each L1 frame). Reconstruction of the packet requires only concatenation of the segments. Depending on their size, a single L1 frame may contain multiple such encapsulated packets or a single portion of a large packet. The L1 frame rate would depend on which L1 logical channel is being used to transport the packets (References [1] and [2]).

6.2.1 PDU Structure

An AAS PDU is contained in an HDLC-like frame delimited by flags as shown in Table 6-1.

Table 6-1: AAS PDU Field Definition

<table>
<tr><th colspan="2">Field</th><th>Bytes</th><th>Description</th></tr>
<tr><td colspan="2">Flag</td><td>1</td><td>0x7E indicates the end of the previous PDU and the start of the current PDU</td></tr>
<tr><td colspan="2">Data Transport Packet Format (DTPF)</td><td>1</td><td>Indicates the format of the current PDU. Refer to Subsection 6.2.3 for details</td></tr>
<tr><td rowspan="3">Conditional</td><td>ECK Length</td><td>1</td><td rowspan="3">The ECK field, in whole, is optional and is only present in specific PDU formats that include restricted access to the AAS data.
These bytes are not used if DTPF = 0x21</td></tr>
<tr><td>ECK No.</td><td>1</td></tr>
<tr><td>ECK Data</td><td>2 to 32</td></tr>
<tr><td colspan="2">PORT</td><td>2*</td><td rowspan="3">AAS Data packets as defined in Section 5</td></tr>
<tr><td colspan="2">SEQ</td><td>2*</td></tr>
<tr><td colspan="2">Payload[ ]</td><td>1 to 2048*</td></tr>
<tr><td colspan="2">Frame Check Sequence (FCS)</td><td>2*</td><td>A 16-bit FCS is used for error detection in little-endian format.</td></tr>
<tr><td colspan="2">Flag</td><td>1</td><td>0x7E indicates the end of the current PDU and the start of the next PDU</td></tr>
</table>

* Payload is limited to a maximum size of 2048 bytes after the addition of any escape bytes as defined in Subsection 6.2.5. The PORT, SEQ, and FCS fields are nominally 2 bytes in length but may be 3 or 4 bytes in length if one or both bytes need to be escaped, respectively.

This PDU structure follows that described in Reference [24] except for the following differences:

1. The Address and Control fields provide no useful function in the HD Radio system and have been eliminated in the interest of efficiency.
2. The Data Transport Packet Format (DTPF) field size is always eight bits
3. No padding is used.
4. The Frame Check Sequence (FCS) is always 16 bits.
5. An ECK (Encryption Control Key) field is conditionally added, depending on the value (format indication) in item 2.

Figure 6-2 shows the structure of the AAS PDU.

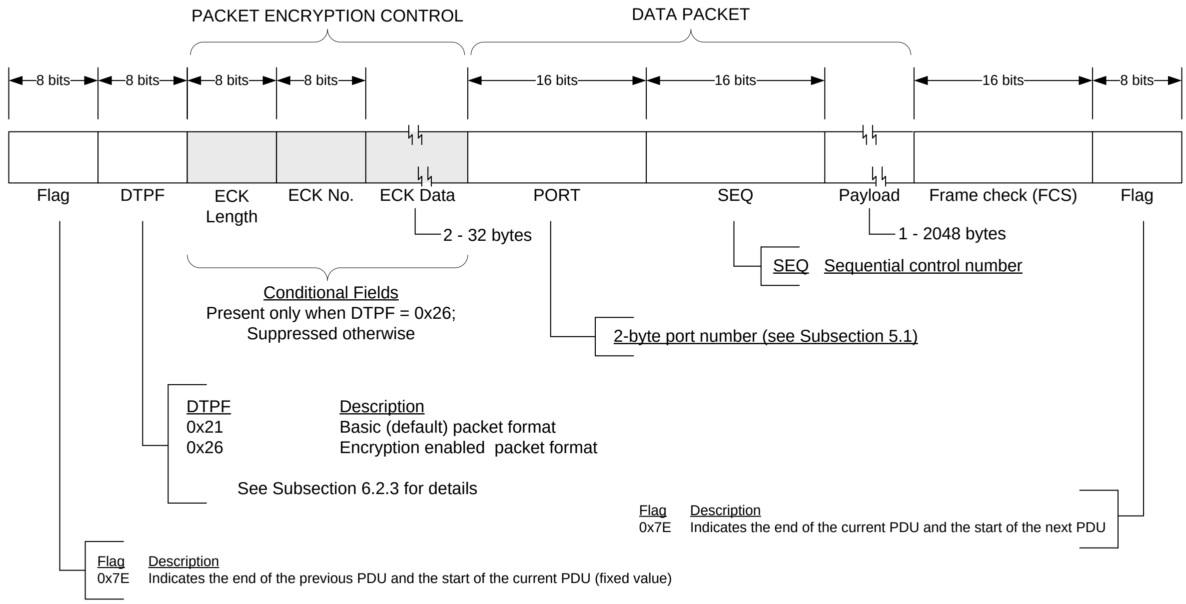

Figure 6-2: AAS PDU Structure

6.2.2 Flag Delimiters

Each PDU is delimited by Flag bytes having the value 0x7E. The Flag delimiters support the following capabilities:

- Only one byte is needed to delimit a packet of any length.
- A false Flag due to a payload error results only in the loss of a single PDU (data packet).
- A corrupted Flag cannot cause a loss of more than two PDUs (data packets).

A single L1 frame may consist of partial or multiple instances of such PDUs. The Flag bytes help in identifying and delimiting each PDU in such instances.

6.2.3 Data Transport Packet Format

The Data Transport Packet Format (DTPF) field, as described in Table 6-2, is used to define the packet format for that PDU. The available (un-used) numbering range allows new packet formats to be added in the future while retaining backward compatibility with older receivers.

Table 6-2: DTPF Allocated Values

DTPF	Description
0x21	Basic packet format. This is the configuration default
0x26	Encryption enabled packet format.
0x00 – 0x20	Reserved for system use
0x22 – 0x25	Reserved for system use
0x27 – 0x7C	Reserved for system use
0x7F – 0xFF	Reserved for system use
0x7D – 0x7E	Invalid. Never used due potential conflict with control flags.

DTPF value 0x21is the default. It indicates that the information contained in the PDU conforms to the basic packet structure defined in Section 5. DTPF value 0x26 indicates that the encapsulated packet format includes additional fields that carry encryption control information. That ECK (Encryption Control Key) includes the key length, key number, and key data for integrity control. All the other DTPF values are unassigned and reserved for future use.

6.2.4 Frame Check Sequence

The Frame Check Sequence (FCS) uses a 16-bit CRC. The FCS is generated using the DTPF field and the fields in the packet structure (refer to Table 5-1) in accordance with Reference [24]. The FCS is in little-endian format.

6.2.5 Transparency

To prevent the values of 0x7E and 0x7D occurring in the data from being read as Flags, an escape mechanism is provided to replace these bytes with a special meaning with alternate values. This is done by replacing the byte with two bytes consisting of the control escape byte 0x7D followed by the original byte exclusive-or’ed with hexadecimal 0x20 (Reference [24]). The only two values that need to be escaped are:

0x7E which is encoded as 0x7D, 0x5E, (Flag Sequence)

0x7D which is encoded as 0x7D, 0x5D (Control Escape)

Since the escape mechanism requires two bytes to encode a single byte, it reduces efficiency by approximately 1% (average) for a long packet with a random data payload.

6.2.6 Idle Pattern

When no packet data is available to send, an idle pattern of repeating Flags may be sent. This is equivalent to a stream of zero length frames and is used so that there is always data to fill a bearer channel.

6.2.7 Application of IETF RFC 1662 for HD Radio Data Transport

Many of the features defined in IETF RFC 1662 (Reference [24]) are inapplicable or unnecessary for data transport in the HD Radio system. In particular, the following sections of the RFC 1662 are ***not*** applicable to the HD Radio system:

- Sections 4.4.2, 4.5.2, and 5 – All streams used for packet transport in the HD Radio system are octet-synchronous.
- Section 6 – No asynchronous-to-synchronous conversion is used.
- Section 7 – The Flag Sequence and Control Escape are the only control flags used in the HD Radio system. Negotiation of additional control characters is not possible and not required.
- Section A – LCP negotiations are not possible and are not used.
- Section B – The PPP frames identified are not valid HD Radio frames.

6.2.8 Encapsulated Packet – Example

Figure 6-3 shows an example of an encapsulated packet (PDU). The payload shown is an ID3 tag for PSD. However, this payload can also be present as AAS Data. As a bearer channel, PSD is also encapsulated in a similar fashion as described in the above sections by the HD RLS mechanism in the PSD Transport. The following elements are noted in the figure:

1. The beginning of the frame is indicated by a Flag Sequence (0x7E).
2. The first byte of the frame is the DTPF field which is set to 0x21, indicating the basic (default) packet format.
3. The next two bytes, following DTPF, contain the Port Number, 0x5100, in little-endian format.
4. A two-byte Sequence Number in little-endian format is shown next. The value of 0x0000 is meaningful only with respect to the sequence numbers of the previous and subsequent packets sent to Port 0x5100.
5. The payload is an ID3 tag that encodes the song title (“Analog Blues”), the artist (“J.Q. Public”), and the album name (“The Lost Sessions”).
6. The payload is followed by a two-byte Frame Check Sequence (FCS) in little-endian format. It is computed over all bytes from the DTPF field through the last byte of the payload.
7. The end of frame is indicated by a Flag Sequence (0x7E).

Note: The byte stream shown could arrive as segments of arbitrary size as long as the byte order is preserved. Lost segments will result in short packets that fail FCS checking.

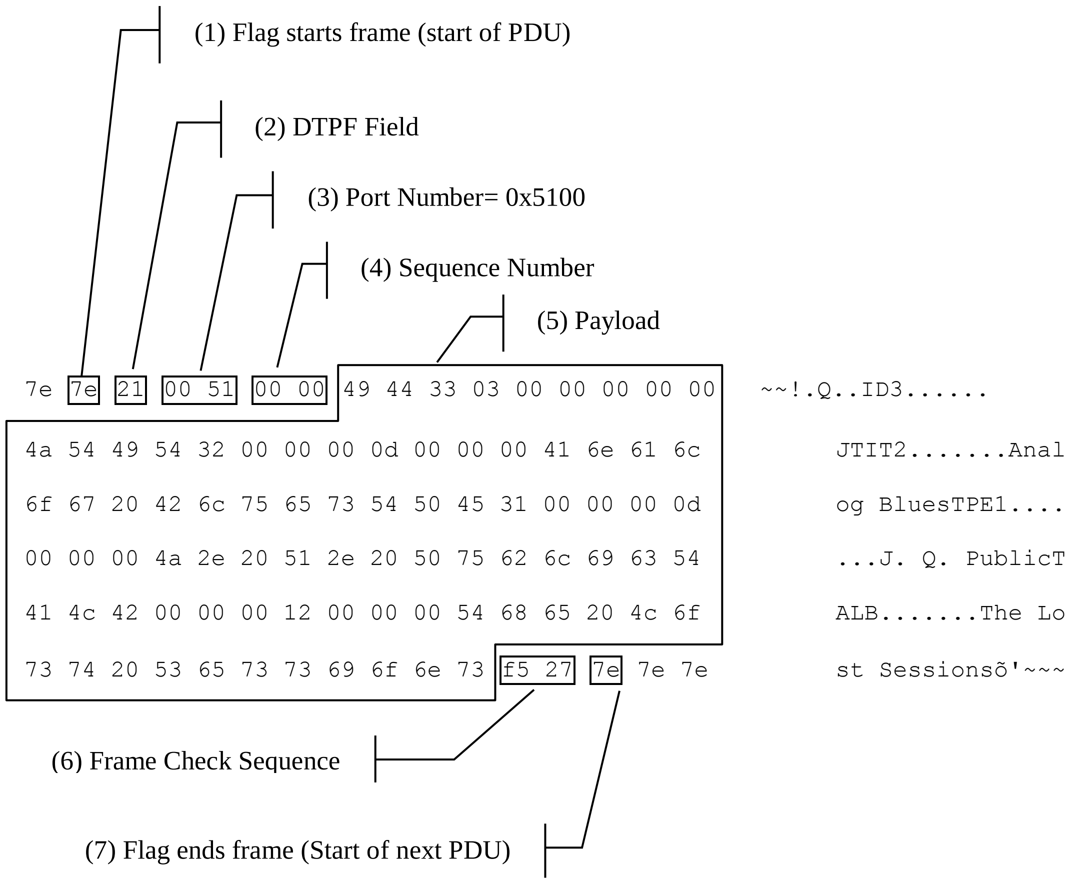

Figure 6-3: Encapsulated Packet – Example

AAS PDUs may span multiple transfer frames as defined in [1] and [2]. Figure 6-4 shows this example. Constraints such as number of streams, size of bearer channel, and others may allow for this scenario. In other scenarios, multiple AAS PDUs may be transmitted within one L1 frame.

L1 Frame 1

Start of AAS PDU	DTPF Field	PORT	Sequence Number	Start of Payload
0x7E	0x21	0xHHHH	0x0001	

L1 Frame 2

Middle section of Payload

L1 Frame 3

End of Payload	FCS	Start Next AAS PDU	DTPF Field	Next PDU
	2 bytes	0x7E	0x21	

Figure 6-4: AAS PDUs Spanning Multiple L1 Frames

6.3 Reed Solomon Coding

All Reed-Solomon (RS) coding in HD RLS uses the extended Galois Field over 2^8 using the characteristics shown below. All codeword blocks are 255 bytes long (that is, no shortened blocks are used). Key characteristics of the Reed-Solomon coding are:

- Primitive polynomial is $x^8+x^4+x^3+x^2+1$ (100011101 in binary notation, most significant bit on the left).
- The RS coding may be configured with any number of roots (parity elements), $p \leq 64$.
- Generator polynomial $g(x) = \prod_{i=1}^{i=p}\left(x - a^i\right)$, where a is a root of the primitive polynomial.
- The first byte of the RS coding block is the leftmost in Figure 6-2. The parity bytes are the rightmost in the payload of each coding block shown in Figure 6-2.

Figure 6-5 shows a block of data, coded using RS (255, 223), which contains 223 data bytes and 32 parity bytes in each block (that is, codeword). The 223-byte data block contains a single "1" followed by 222 zeros. The 32 RS parity bytes are located at the end of the block and are shaded in gray.

1	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	8b
1b	e9	a3	e3	cb	72	1b	ba	1c	2e	5c	06	8b	93	b1	03
93	37	e7	b7	d4	ca	e3	61	9c	f4	e1	de	74	8d	f3	

Data Bytes

RS Parity Bytes

Figure 6-5: Reed-Solomon (255, 223) Coding Example

6.3.1 Coding Rates

The AAS configuration mechanism supports a wide range of RS encoder configurations, resulting in a wide range of coding rates. The encoder may be configured from (255,253) to (255,191).

Due to practical considerations, such as simplified configuration control and reduced throughput versus error correction benefits, the following configuration is recommended: (255,223), thus providing a coding rate of r = 0.875.

It is noted that the recommended configuration applies only if and when FEC is applied.

6.4 Byte Interleaving

6.4.1 Overview

The HD RLS mechanism employs a convolutional byte interleaver. An interleaver row consists of 255 columns (bytes), which is the size of one RS codeword. Thus, the FEC stream is a series of 255-byte RS encoded blocks, and the interleaver design allows decoding to start at any FEC block in the sequence. The interleaver column size depends on the error protection level.

The RS codeword bytes are mapped to the interleaver matrix. Write operations are sequential in rows, while read operations are addressed by the applicable equations given in Subsections 6.4.2.1 and 6.4.2.2. Every (entire) codeword is written into a row, starting from the first row and first column, so that the rows and the codewords are aligned. Consecutive codewords are placed in consecutive rows.

6.4.2 Interleaver Equations

6.4.2.1 Interleaver Column Selection – Read Operation

Apply the following equation using the applicable parameters:

$$Column(i) = (i \cdot N_s) MOD(255)$$

where N_s is 53

6.4.2.2 Interleaver Row Selection – Read Operation

Apply the following equation using applicable parameters:

$$Row(i) = [i - 254 \cdot INT(i/255)] MOD(R_w)$$

where R_w is the number of rows in the interleaver matrix

6.4.3 Interleaver Timing

The interleaving operation starts by writing to the first location of the interleaver, marked as (0, 0). After the first write operation, every write operation has to be followed by a read operation. This means that the first location of each row is being written into and read out immediately, experiencing no delay.

The process continues, while the write operations are by row and the read operations are by using the equations above.

6.4.4 Interleaving Range

The interleaver may be configured to a span (“depth”) ranging from zero code words (meaning no interleaving) to 64 code words. The default operation setting is “no interleaving.”

6.5 Block Synchronization

FEC and interleaving occur after every *n* blocks of 255 bytes (the length of a Reed Solomon codeword). The exact number *n* depends on the specific configuration, as further discussed. To enable the receiver to identify the start of these blocks, a four-byte Block Boundary Marker (BBM) is regularly inserted into the stream to indicate the start of a block, as shown in Figure 6-6.

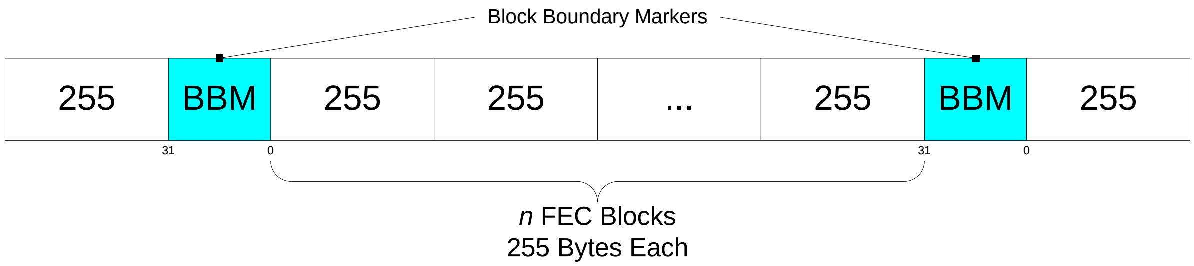

Figure 6-6: FEC Blocking with Block Boundary Markers

The number of FEC blocks between BBMs is set at *n=1* for Opportunistic bearer channels, and *n=4* for Fixed bearer channels.

The BBM is defined as follows,

BBM = [01111101001110101110001001000010]

where bit 0 (least significant bit (LSB)) is on the right and the left-most bit is sent first.

7 Bearer Channels

The HD Radio broadcast system supports multiple bearer channels to transport packets in one or more encoded byte streams. The AAS PDUs are transmitted on the fixed and opportunistic bearer channels and the PSD PDUs are transmitted on the PSD bearer channel. The HD RLS enables the encapsulation of the data packets (both AAS and PSD) as PDUs before they are sent over the different bearer channels to Layer 2 for further processing.

7.1 Layer 2 PDU Packing

Figure 7-1 shows how the digital audio and data are packed into a Layer 2 PDU. In addition to transporting the data, Layer 2 also provides an indication whether the Layer 2 PDU contains audio, opportunistic data, and/or fixed data.

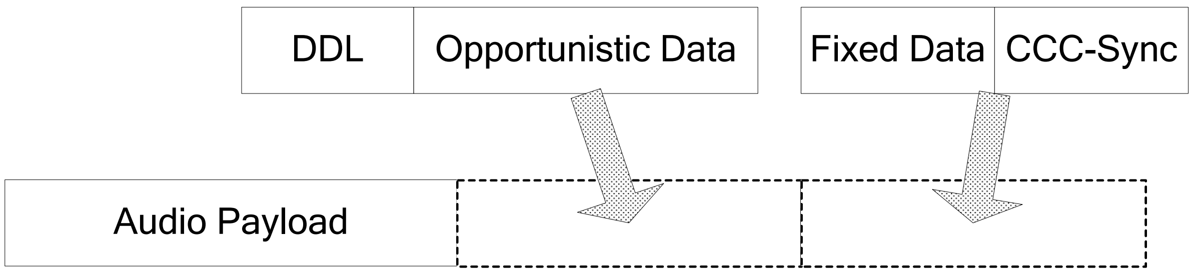

Figure 7-1: Layer 2 PDU Packing

Reference [3] provides a detailed description of the Layer 2 transmit processing.

7.2 Bearer Channel Comparison

Table 7-1: Bearer Channel Comparison

Bearer Channel Type	Bearer Channel Description	Reed Solomon	Interleaver	BBM Frequency
PSD	Bytes allocated in MPS/SPS PDU	FEC not used		
OPPORTUNISTIC	Unused bytes allocated to audio programs Variable capacity	(255,223)	None	1 : 1
FIXED	Uses allocated segment(s) of L2 frame “Infinitely” variable FEC	(255,255) to (255,191)	0 to 64 Blocks	1 : 4

7.3 PSD Bearer Channel

A PSD bearer channel uses bytes allocated within the Audio Transport for the transmission of program-related packet data. The encoded byte stream transmitted over a PSD bearer channel contains encapsulated packets (PSD PDUs) from the PSD Transport with no forward error correction. The Audio Transport provides the mechanism to identify the number and position of the encoded bytes in each transmitted Audio Transport PDU.

7.4 Opportunistic Bearer Channel

During silence and simple audio passages the encoded digital audio might require less than its allocated bandwidth. When this occurs, the unused capacity may be used to send data packets over the opportunistic bearer channel. The size of the opportunistic payload is determined on the basis of whether the audio programs use their full allocated capacity or not.

The data packet encapsulation is described in Subsection 6.2.

The FEC chain is defined in Section 7.4.1 and the settings are not configurable.

7.4.1 FEC Coding for Opportunistic Bearer Channels

For opportunistic bearer channels the byte stream uses:

- A fixed Reed Solomon coding rate of (255,223)
- No byte interleaving
- A block boundary marker (BBM) for every Reed Solomon codeword

This choice of settings provides robust error tolerance and minimal latency.

7.4.2 Opportunistic Data Identification/Delimiting

To allow opportunistic data to be identified in the Layer 2 PDU (Reference [3]), a five-byte data delimiter (DDL) field is used to identify the start of the opportunistic data in the PDU. The end of the opportunistic data is either the end of the PDU or the start of the fixed data payload (if present).

The five-byte DDL sequence has a binary value of [1001110100101111001110010000010111011000] where ddl_0 (LSB) is on the right.

7.5 Fixed Bearer Channel

A fixed bearer channel contains the following:

- A Synchronization Channel (SYNC)
- A Configuration Control Channel (CCC)
- From one to four sub-channels containing fixed byte streams with different FEC

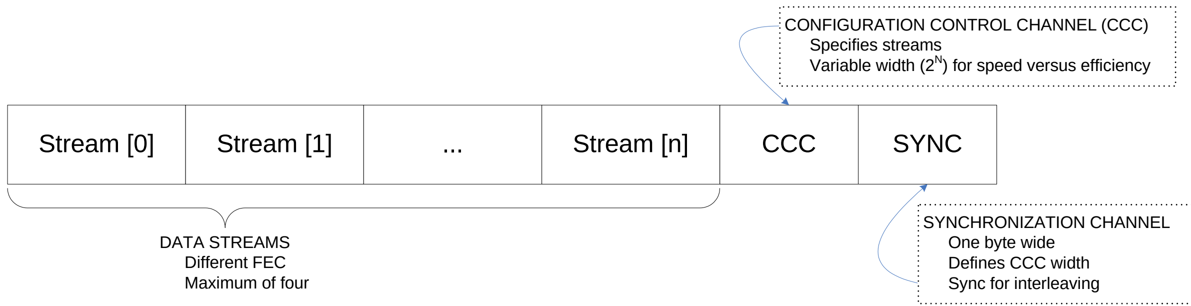

Figure 7-2: Fixed Bearer Channel Structure

7.5.1 Synchronization Channel

The one-byte wide Synchronization Channel, as described in Table 7-2, encodes the Configuration Control Channel width and timing information to allow AAS PDU synchronization between the transmitter and receiver. The bytes are sent in the following sequence: count, width, width, width; count, width, width, width; count, width, width, width; and, the pattern continues.

Table 7-2: Synchronization Channel

Synchronization Channel		
Element	**Type**	**Description**
width	UINT8	Valid values for the Configuration Control Channel sequence width (in bytes) are any even number from 2 through 30, or one. These values are represented in the Synch Width byte in the form 0xNN, where N is one-half the number of bytes of the Configuration Control Channel. The byte 0xNN consists of two four-bit nibbles, each of the value 0xN. Thus, for N=0x1 through 0xF, the Configuration Control Channel width takes the value of an even number from 2 through 30 bytes (0xN · 2) bytes. For example, 0x44 indicates a CCC width of 8 bytes. For N=0, a CCC width of one byte is defined.
count	UINT8	The initial value of the count is zero. Each time the count is sent, it is incremented by 4 modulo 256. The count is intended to allow for synchronizing fixed channel reconfiguration in the future.

7.5.2 Configuration Control Channel

The Configuration Control Channel (CCC), as described in Table 7-3, sends a repeating message describing the number, width, and FEC configuration of the fixed sub-channels. The Configuration Control Channel can describe from one to four fixed sub-channels using from one to 20 bytes to allow for the optimum compromise between channel overhead and the time to process a complete fixed configuration message. The message is encapsulated to provide framing and error detection. Note that the mode, length, and CRC-16 elements may need additional escape bytes to be applied, as defined in Subsection 6.2.5. However, these escape bytes are not counted in the computation of the Synchronization Channel width element.

Table 7-3: Configuration Control Channel

HD RLS Configuration Control Channel Message		
Element	**Type**	**Description**
0x7E	UINT8	HDLC Flag When computing the value of the Synchronization Channel width element, this byte is counted
MPL	UINT8	Byte to indicate the maximum packet length of each of the four sub-channels. See Subsection 7.5.2.1 for details.
mode[0]	UINT16	A 16-bit value indicating the FEC encoding of the 0^{th} sub-channel
length[0]	UINT16	A 16 bit value indicating the number of bytes in the 0^{th} sub-channel (in little-endian format)
…		Additional mode and length pairs
mode[n]	UINT16	Mode of n^{th} sub-channel, where n has a maximum value of 3
length[n]	UINT16	Length of n^{th} sub-channel in bytes (in little-endian format)
CRC-16	UINT16	16 bit FCS supplied by HDLC provides error detection
0x7E	UINT8	HDLC Flag for next message

The mode bits are defined in Figure 7-3. The default value of 0x0000 indicates no FEC encoding or interleaving.

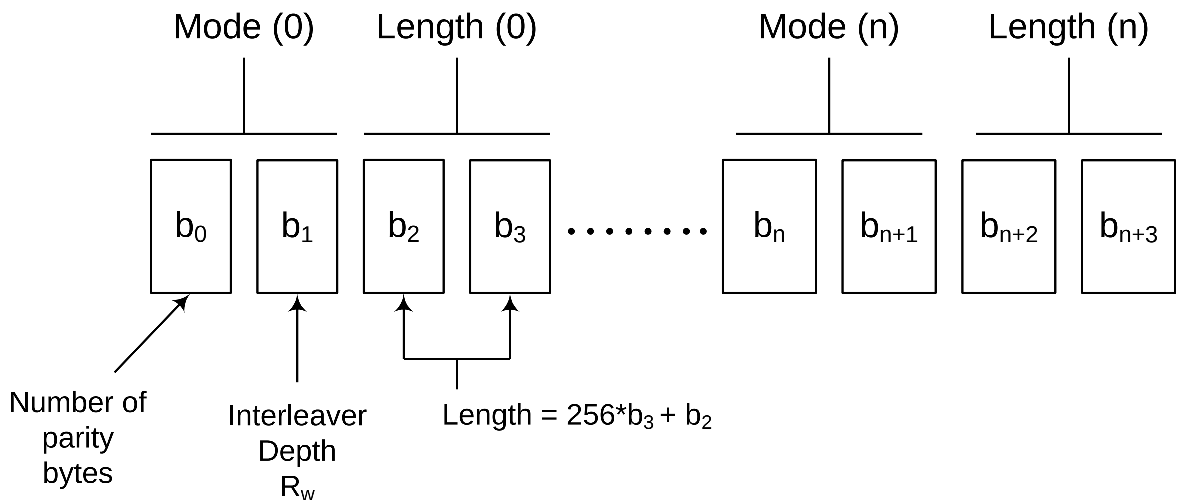

Figure 7-3: Mode Bit Definitions

In Figure 7-3, b_0 and b_1 are the high-byte and low-byte of Mode (0); b_2 and b_3 are the high-byte and low-byte of Length (0).

Where,

- b_0 = Number of Parity Bytes, N_p – This is the number of parity bytes in each Reed-Solomon codeword. A value of N_p implies a coding of (255:255-N_p). The maximum value of N_p is 64.
- b_1 = Interleaver Depth – Number of interleaver rows, R_w. The maximum value is 64.
- b_2 and b_3 = Length (in little-endian format) – The number of bytes the sub-channel is allocated in each Layer 2 PDU
-

A fixed portion of the Layer 2 PDU is allocated for every fixed sub-channel byte stream. The length – b2 and b3 – refers to the actual message length of the fixed sub-channel byte stream in the Layer 2 PDU structure.

The Configuration Control Channel messages require two HDLC flags to delimit them. However, as noted in Table 7-2, valid values for the CCC width are any even number from 2 through 30 bytes, or one byte. Therefore, additional dummy bytes are padded in order to maintain the even width and the byte counting as shown in Table 7-3.

The minimum number of bytes needed to transmit a single Configuration Control Channel message is eight: 1 flag, 1 padding byte, mode, length, and CRC-16. Therefore, a one-byte Configuration Control Channel message can specify a single fixed sub-channel payload configuration in 8 L1/L2 PDU structures. An 8-byte wide Configuration Control Channel accomplishes this in a single L1/L2 Frame PDU structure.

The fixed bearer channel Configuration Control Channel can describe from one to four fixed sub-channels. The total required number of Configuration Control Channel bytes is listed in Table 7-4.

Table 7-4: Required CCC Length

Number of Sub-Channels	CCC Total Length [bytes]
1	8
2	12

Number of Sub-Channels	CCC Total Length [bytes]
3	16
4	20

7.5.2.1 Maximum Packet Length (MPL) Indicator

The control channel includes a byte to signal to the receiver how much memory to allocate for data packets in each of the four sub-channels as follows:

MPL bits 7, 6 - maximum packet length in sub-channel 3

MPL bits 5, 4 - maximum packet length in sub-channel 2

MPL bits 3, 2 - maximum packet length in sub-channel 1

MPL bits 1, 0 - maximum packet length in sub-channel 0

Refer to Table 7-5 for a definition of each bit pair.

Table 7-5: Definition of MPL Bit-Pairs

Most Significant Bit (7, 5, 3, or 1)	Least Significant Bit (6, 4, 2, or 0)	Description
0	0	Maximum packet size (*) of sub-channel n is 4096 bytes (**)
0	1	Maximum packet size (*) of sub-channel n is 2048 bytes
1	0	Maximum Packet size (*) of sub-channel n is 1024 bytes
1	1	Maximum Packet size (*) of sub-channel n is defined by length[n] parameter in Control Channel. Receiver will limit this value to 4096 bytes (**) maximum

Notes:

* Packet Size is defined as the length of the encapsulated payload as defined in Table 6-1. It includes escape bytes.

** Although the maximum packet size is defined as 2048 bytes in this document, a value of 4096 may be used in legacy systems. However, if a value of 4096 is indicated, byte interleaving as defined in Subsection 6.4 is not permitted: i.e., the mode bytes in the Control Channel must indicate an interleaver depth of zero for all sub-channels.

7.5.3 Fixed Sub-Channel Byte Stream

Each fixed sub-channel byte stream uses:

- A fixed Reed Solomon coding rate of (255, 255-N_P), where N_P is the number of parity bytes specified in the configuration message and $N_P \leq 64$.
- Byte interleaver depth of N_I, where N_I is the length of the interleaver in units of 255-byte blocks. $N_I \leq 64$ and is specified in the configuration message.
- A block boundary marker (BBM) for every four FEC block codewords.

Figure 7-4 shows an example of the fixed sub-channel byte stream for a 32-bit Reed Solomon parity.

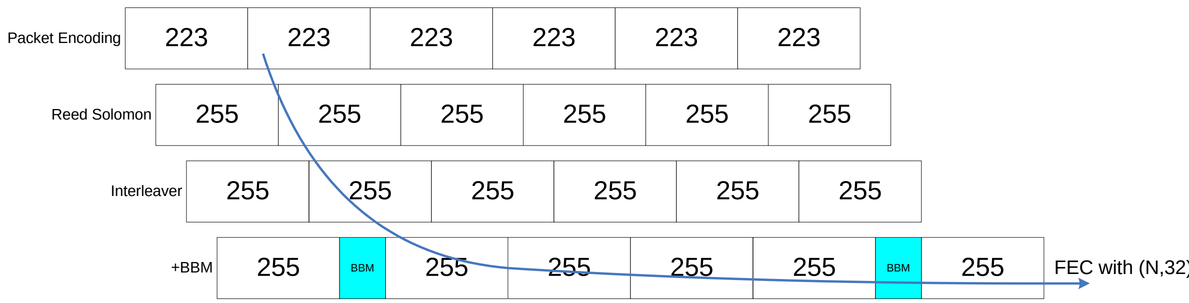

Figure 7-4: Fixed Sub-Channel Byte Stream

Transmission Signal Quality Metrics for FM IBOC Signals

Version 03

November 2022

SY_TN_2646s

TRADEMARKS

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1 Abstract

This technical report describes the measurement of several transmission signal quality metrics specified in Reference [7] for FM In-Band-On-Channel (IBOC) signals defined in Reference [1]. In particular, the metrics are intended to assess signal distortion due to both nonlinear high-power amplifier (HPA) characteristics as well as linear filtering. The effects of the Peak-to-Average Power Ratio (PAPR) reduction algorithm on the reference subcarriers and the data (non-reference) subcarriers are measured with different metrics, as appropriate. Group delay and gain variation across subcarriers may also be measured with the techniques described.

This document also defines transmission signal quality metrics for the quadrature amplitude modulation (QAM) waveforms used in advanced primary service modes, as well as for the QPSK waveform used in secondary service mode MS5.

NOTE

> Version 02, SY_TN_2646s - Transmission Signal Quality Metrics for FM IBOC Signals
>
> Legacy Appendix B showcased constellation plots and plots of MER versus partition index/subcarrier index.
>
> These legacy plots have been removed from this document.

2 Acknowledgements

The selection of Modulation Error Ratio (MER) as the HD Radio transmission signal quality metric for the FM IBOC signal and the method of measuring MER on the FM IBOC signal was developed by a working group of technologists representing iBiquity Digital, Broadcast Electronics, Continental Electronics, Harris Broadcast, Nautel Ltd., and other interested participants from the US radio broadcast industry. This group of technologists reached full consensus on the standardized method for FM IBOC signal MER measurement described in this document.

The list of regular participants included:

Geoffrey Mendenhall, Working Group Chairman - Harris Broadcast

Kevin Berndsen – Harris Broadcast

Harvey Chalmers – iBiquity Digital

Jeff Detweiler – iBiquity Digital

David Gates – Cesium Communications

Tim Hardy – Nautel Ltd.

Dave Hershberger – Continental Electronics

John Kean – NPR Laboratories

Brian Kroeger – iBiquity Digital

Dave Kroeger – Broadcast Electronics

David Layer – NAB Science & Technology

David Maxson – Broadcast Signal Lab

Phillip Schmid – Nautel Ltd.

Burt Weiner – Burt I. Weiner Associates

A special acknowledgement is extended to Kevin Berndsen, Harvey Chalmers, Dave Hershberger, Brian Kroeger, and Phillip Schmid for doing the majority of the work to develop, test, and verify the HD Radio MER measurement method described in this document.

3 References

STATEMENT
Each referenced document that is mentioned in this document shall be listed in the following iBiquity document:

- Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard Document Number: SY_REF_2690s

4 Abbreviations, Symbols, and Acronyms

Abbreviations, Symbols, Acronyms

BPSK	Binary Phase Shift Keying
dB	decibel
EVM	Error Vector Magnitude
FFT	Fast Fourier Transform
FM	Frequency Modulation
HPA	High Power Amplifier
Hz	Hertz
I	In-Phase
IBOC	In-Band-On-Channel
L1	Layer One
MER	Modulation Error Ratio
NRSC	National Radio Systems Committee
OFDM	Orthogonal Frequency Division Multiplexing
PAPR	Peak-to-Average Power Ratio
PLL	Phase-Locked Loop
PPM	parts per million
Q	Quadrature
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RF	Radio Frequency
rms	Root Mean Square

5 Measurement Conditions

Prior to OFDM demodulation, it is first necessary to perform frequency and symbol synchronization commonly across all subcarriers. A suggested method to accomplish this is described in Appendix A.

The sample rate (nominally 744187.5 complex samples per second) should be locked to 2160 times the symbol rate (nominally 344.53125 Hz). It is important that the symbol synchronization is accurate, since any symbol timing error will cause a linear phase shift across the subcarriers. However, group delay variations across the subcarriers will prevent perfect timing for every subcarrier. Further timing correction for individual subcarriers should not be implemented since this is an error we want to measure. Of course, both frequency and symbol synchronization algorithms already exist for receivers. For the purposes of these static measurements in non-real time, a block of at least 128 OFDM symbols can be used to estimate the frequency and symbol timing parameters as a block (instead of PLLs, etc.), simplifying the synchronization process.

The first step is to capture N contiguous symbols of 2160 complex baseband samples each, where N is the block size. For good results, N should be 128 or greater. A value of 512 symbols corresponds to a total time span of one L1 Frame period (1.486 seconds).

A cyclic folding operation with root-Nyquist pulse-shape weighting should be applied to each OFDM symbol, converting the 2160 complex samples into 2048. Next, a 2048-point CFFT should be computed for each of the N symbols. The bins of the CFFT correspond to the OFDM subcarriers.

Some of the transmission signal quality measurements specified in Reference [7] may be conveniently computed using the reference subcarriers. These measurements include gain variation, group delay variation and error vector magnitude (EVM). However, in place of EVM, a slight variant called modulation error ratio (MER) is proposed in order to be in better alignment with standard practices within the broadcast industry.

The subcarrier index m corresponds to the active reference subcarriers, where $m \in \{-546, -527, -508, -489, -470. -451, -432, -413, -394, -375, -356, -337, -318, -299, -280$, -267, -248, -229, -210, -191, -172, -153, -134, -115, -96, -77, -58, -39, -20, -1, 1, 20, 39, 58, 77, 96, 115, 134, 153, 172, 191, 210, 229, 248, 267, 280, 299, 318, 337, 356, 375, 394, 413, 432, 451, 470, 489, 508, 527, 546$\}$. The active subcarriers per service mode are given in Table 5-1. Refer to [1] for further details. All processing of reference subcarriers will be similar, whether on the upper or lower sidebands, on the primary or secondary sidebands, or bounding partitions modulated with QPSK, 16-QAM, or 64-QAM.

Note: There are no data subcarriers between reference subcarriers (-280, -267), (-1, 1), and (267, 280).

Table 5-1: Active Reference Subcarriers per Service Mode

Service Mode	Active Reference Subcarriers	
	Lower Sideband	**Upper Sideband**
MP1	−546, −527, −508, −489, −470, −451, −432, −413, −394, −375, −356	356, 375, 394, 413, 432, 451, 470, 489, 508, 527, 546
MP2	−546, −527, −508, −489, −470, −451, −432, −413, −394, −375, −356, −337	337, 356, 375, 394, 413, 432, 451, 470, 489, 508, 527, 546
MP3	−546, −527, −508, −489, −470, −451, −432, −413, −394, −375, −356, −337, −318	318, 337, 356, 375, 394, 413, 432, 451, 470, 489, 508, 527, 546
MP5, MP6, MP11, MP1X, MP1XOV, MP6OV, DSB1, DSB1OV	−546, −527, −508, −489, −470, −451, −432, −413, −394, −375, −356, −337, −318, −299, −280	280, 299, 318, 337, 356, 375, 394, 413, 432, 451, 470, 489, 508, 527, 546
MS5	-267, -248, -229, -210, -191, -172, -153, -134, -115, -96, -77, -58, -39, -20, -1	1, 20, 39, 58, 77, 96, 115, 134, 153, 172, 191, 210, 229, 248, 267

6 Phase Adjustment for each Reference Subcarrier

The average phase θ in radians (with ambiguity of π radians) of each reference subcarrier is computed as the argument (-π to π) of the sum of the squared complex reference subcarrier values (where the complex reference subcarrier values are denoted by *r*), divided by 2. The division by 2 is necessary because the squaring of the values doubles the phase. This phase represents the phase of the BPSK symbol constellation point. The symbol within the block of *N* consecutive symbols is represented by *n*, while *m* represents the reference subcarrier.

$$\theta_m = \frac{1}{2}\arg\left\{\sum_{n=0}^{N-1} r_{n,m}^2\right\} \; ; \; -\frac{\pi}{2} \le \theta_m < \frac{\pi}{2}.$$

A sample rate (frequency) error between the exciter clock and the measurement equipment will result in a linear phase ramp over the time span for any one subcarrier. The span of this ramp is proportional to the sample error drift over the signal duration. The slope of the ramp is computed for each subcarrier index *m*.

$$slope_m = \frac{d\theta_{n,m}}{dn} = \frac{1}{2}\cdot\arg\left\{\sum_{n=1}^{N-1}\left(r_{n,m}\cdot r_{n-1,m}^*\right)^2\right\} \;\; ; \;\; -\frac{\pi}{2} \le slope_m < \frac{\pi}{2}$$

The phase can now be estimated for each OFDM symbol, and for each reference subcarrier.

$$\theta_{n,m} = \theta_m + slope_m \cdot \left(n - \frac{N-1}{2}\right)$$

For subsequent magnitude and noise computations, the BPSK symbols of each reference subcarrier are phase-adjusted to place the nominal BPSK constellation points on the real axis.

$$u_{n,m} = r_{n,m} \cdot e^{-j\cdot\theta_{n,m}}$$

For these cases it is assumed that the reference subcarriers are not processed by the PAPR reduction algorithm, and the distortion of I and Q is small at the exciter output. However, some versions of the PAPR reduction algorithm allow some amplitude variation of the reference subcarriers, although well within acceptable noise-level limits. This is primarily a measure of the signal distortion caused by the path (e.g., HPA and filters) between the exciter output and the antenna.

The above algorithms should work well for a sample clock frequency error of approximately ±2.5 PPM for a block size of N=512. The tolerance is inversely proportional to the block size so that if N=128, a tolerance of ±10 PPM is acceptable. It may not be necessary to compute the phase slope when the samples are perfectly synchronized within the exciter (*slope* = 0).

7 Signal Magnitudes and Gain Deviation

The magnitude of each BPSK reference subcarrier m can now be conveniently computed as the mean of the absolute value of the real part of the phase-adjusted symbols.

$$smag_m = \frac{1}{N} \cdot \sum_{n=0}^{N-1} \left| \mathrm{Re}\{u_{n,m}\} \right| \ ;$$

where N = the total number of OFDM symbols within the current block of data and m represents an active reference subcarrier index.

The gain variation across the upper or lower primary subcarriers is determined by

$$\Delta GpdB_{upper} = 20 \cdot log\left(\frac{max(smag)}{min(smag)}\right);\ \text{for } m \geq 280$$

$$\Delta GpdB_{lower} = 20 \cdot log\left(\frac{max(smag)}{min(smag)}\right);\ \text{for } m \leq -280$$

The gain variation across the upper or lower secondary subcarriers is determined by

$$\Delta GsdB_{upper} = 20 \cdot log\left(\frac{max(smag)}{min(smag)}\right);\ \text{for } 1 \leq m \leq 267$$

$$\Delta GsdB_{lower} = 20 \cdot log\left(\frac{max(smag)}{min(smag)}\right);\ \text{for } -267 \leq m \leq -1$$

The above computations are a convenient way to verify the FM in-band on-channel gain flatness specification [7], Subsection 4.9.

8 Group Delay

The local group delay over each active partition *m* can be approximated over a finite frequency difference between any two adjacent reference subcarriers indexed *m* and *m*-19 (19 subcarrier spacing) for the upper partitions, and reference subcarriers indexed *m* and *m*+19 for the lower partitions. The average phase θ_m for each reference subcarrier is used for this computation. Notice that the innermost reference subcarrier indices are not used to identify a partition, since the partition is identified by the index of the outermost Reference Subcarrier bounding the partition. Also, there is no partition between primary and secondary subcarriers. The ambiguity is resolved when the phase difference exceeds π/2.

$$\tau\_upper_{m,} = \frac{\theta_{m-19} - \theta_m - \pi \cdot round\left(\frac{\theta_{m-19} - \theta_m}{\pi}\right)}{2 \cdot \pi \cdot 19 \cdot 344.53125 \cdot 135/128} \quad \text{seconds,}$$

or equivalently,

$$\tau\_upper_{m,} = 23052 \cdot \left(\theta_{m-19} - \theta_m - \pi \cdot round\left(\frac{\theta_{m-19} - \theta_m}{\pi}\right)\right) \text{ nanoseconds };m > 0$$

$$\tau\_lower_m = \frac{\theta_m - \theta_{m+19} - \pi \cdot \left(round\left(\frac{\theta_m - \theta_{m+19}}{\pi}\right)\right)}{2 \cdot \pi \cdot 19 \cdot 344.53125 \cdot 135/128} \quad \text{seconds,}$$

or equivalently,

$$\tau\_lower_m = 23052 \cdot \left(\theta_m - \theta_{m+19} - \pi \cdot \left(round\left(\frac{\theta_m - \theta_{m+19}}{\pi}\right)\right)\right) \text{ nanoseconds}; m < 0$$

The maximum group delay variation is estimated by

$$\Delta\tau\_upper = \max(\tau\_upper) - \min(\tau\_upper)$$

$$\Delta\tau\_lower = \max(\tau\_lower) - \min(\tau\_lower),$$

where the max and min values are obtained independently for all values of m.

The above computations are a convenient way to verify the FM in-band on-channel group delay flatness specification [7], Subsection 4.10.

Any additional group delay caused by symbol synchronization timing error will not affect this difference. Note that the specification is equivalent to 0.75 degrees over a frequency span between the two reference subcarriers with the minimum and maximum group delay.

9 Reference Subcarrier MER

The rms noise may be computed for each (phase-adjusted) reference subcarrier. This noise voltage is normalized to the subcarrier nominal magnitude.

$$Vnref_m = \frac{1}{smag_m} \cdot \sqrt{\frac{1}{N} \cdot \sum_{n=0}^{N-1} \left[\left(\left| Re\{u_{n,m}\} \right| - smag_m \right)^2 + Im\{u_{n,m}\}^2 \right]}$$

The normalized rms noise is then converted to MER:

$$MERref_m = -20 \cdot \boldsymbol{log_{10}}(\, Vnref_m \,),$$

or equivalently:

Equation 1: MER per Reference Subcarrier

$$MERref_m = -10 \cdot log_{10} \left(\frac{1}{N \cdot smag_m^2} \cdot \sum_{n=0}^{N-1} \left[\left(\left| Re\{u_{n,m}\} \right| - smag_m \right)^2 + Im\{u_{n,m}\}^2 \right] \right)$$

Equation 1 computes the MER for each individual reference subcarrier. To get a single signal quality metric for all upper or lower Primary reference subcarriers, the MER values can be averaged according to Equations 2(a) and 2(b). To get a single signal quality metric for all upper or lower Secondary reference subcarriers, the MER values can be averaged according to Equations 2(c) and 2(d). *Mref* is the number of active reference subcarriers and *Mstart* is the index to use for the summation (see Table 9-1).

Equation 2: Composite MER Measurements for Upper and Lower Reference Subcarriers

2(a) $$MER_{pref\_upper} = 10 \bullet \log_{10}\left(\left(\frac{1}{Mref}\right) \bullet \sum_{m=0}^{Mref-1} 10^{\left(\frac{MERref_{(Mstart+m\cdot 19)}}{10}\right)}\right)$$

2(b) $$MER_{pref\_lower} = 10 \bullet \log_{10}\left(\left(\frac{1}{Mref}\right) \bullet \sum_{m=0}^{Mref-1} 10^{\left(\frac{MERref_{(Mstart-m\cdot 19)}}{10}\right)}\right)$$

2(c) $$MER_{sref\_upper} = 10 \bullet \log_{10}\left(\left(\frac{1}{Mref}\right) \bullet \sum_{m=0}^{Mref-1} 10^{\left(\frac{MERref_{(Mstart+m\cdot 19)}}{10}\right)}\right)$$

2(d) $$MER_{sref\_lower} = 10 \bullet \log_{10}\left(\left(\frac{1}{Mref}\right) \bullet \sum_{m=0}^{Mref-1} 10^{\left(\frac{MERref_{(Mstart-m\cdot 19)}}{10}\right)}\right)$$

Table 9-1: Number of Active Reference Subcarriers (Mref) and Summation Index (Mstart) versus Service Mode

Service Mode	*Mref*				*Mstart*	
	Primary		Secondary		Lower Sideband	Upper Sideband
	Lower	Upper	Lower	Upper		
MP1	11	11	---	---	-356	356
MP2	12	12	---	---	-337	337
MP3	13	13	---	---	-318	318
MP5	15	15	---	---	-280	280
MP6	15	15	---	---	-280	280
MP11	15	15	---	---	-280	280
MP1X	15	15	---	---	-280	280
DSB1	15	15	---	---	-280	280
MP1XOV	15	15	---	---	-280	280
MP6OV	15	15	---	---	-280	280
DSB1OV	15	15	---	---	-280	280
MS5	---	---	15	15	-1	1

In Equations 2(a) through 2(d), the sums are computed for the active reference subcarriers in the Primary/Secondary upper and lower sidebands, respectively. The variable *Mref* equals the total number of active reference subcarriers on one Primary or Secondary sideband (upper or lower).

It should be noted that the composite MER measurements computed by Equations 2(a), 2(b), 2(c), and 2(d) exclude any group delay or amplitude variations across the channel bandwidth. This is because each reference subcarrier has been separately adjusted for amplitude and phase variations as described in Sections 6 and 7. This is appropriate since the receiver will also process each reference subcarrier separately. Amplitude and group delay distortions are specified separately in [1] and do not need to be included in the MER.

In addition, the reference subcarriers do not have the PAPR reduction algorithm applied. So the reference subcarrier MER is a good measure of the signal constellation spread caused by noise and distortion introduced solely by the transmission signal path after the digital-to-analog conversion in the exgine/exciter.

10 PAPR Reduction Noise on Data Subcarriers

The PAPR Reduction algorithm introduces noise onto the data-bearing, non-reference subcarriers. Since these noise samples are intentionally biased in a way that mitigates errors, a simple rms or MER computation is not appropriate here. The measurement strategy is to first adjust the symbol constellations of the data subcarriers such that the constellation points are contained within their appropriate cells and the in-phase (I) and quadrature (Q) components of each point is nominally centered within that cell. The metric in this case would penalize values that are closer to the decision region axes.

11 Equalization of Data Subcarriers

The data subcarriers within each partition of 18 subcarriers bounded by reference subcarriers on either side must be equalized to perform subsequent measurements. A linear interpolation of the outer reference subcarriers is used to equalize each data subcarrier across each OFDM symbol. The reference phases are first adjusted to avoid complications due to modulo π wrapping.

$$\text{If } \left|\theta_{n,m} - \theta_{n,m+19}\right| > \frac{\pi}{2}\text{ , then let } \theta_{n,m+19} = \theta_{n,m+19} + \pi \;;$$

where n is the OFDM symbol index, m is the index of the active reference subcarrier to the left (lower index number) of the partition, $\theta_{n,m}$ and $\theta_{n,m+19}$ are the phases of the corresponding reference subcarriers that bound the partition, and $\theta_{n,m+19}$ may be adjusted for the modulo π wrap case.

Next, compute a set of 18 equalizer coefficients to be applied to each of the 18 data subcarriers (k=1,2,…18). A phase shift of π/4 (equivalently a factor of 1+j) is also applied to center the constellations in their appropriate cells. The equalizer coefficients for the partition, referenced by outermost frequency reference subcarrier *m that bounds the partition,* are computed as

$$C_{n,k,m} = \frac{19\cdot(1+j)}{(19-k)\cdot smag_m \cdot e^{j\cdot\theta_{n,m}} + (k)\cdot smag_{m-19}\cdot e^{j\cdot\theta_{n,m-19}}}\;;k=1,2..18\;;\; m>0$$

$$C_{n,k,m} = \frac{19\cdot(1+j)}{(19-k)\cdot smag_m \cdot e^{j\cdot\theta_{n,m}} + (k)\cdot smag_{m+19}\cdot e^{j\cdot\theta_{n,m+19}}}\;;k=1,2..18\;;\; m<0$$

where k is the data subcarrier within the partition.

Notice that there are no partitions corresponding to reference subcarrier indices -280, -1, 1, and 280.

Then apply the equalizer coefficients to the data subcarrier symbols within each partition.

$$v_{n,m-k} = s_{n,m-k}\cdot C_{n,k,m}\,;k=1,2,\ldots18\,;\text{for } m>0$$

$$v_{n,m+k} = s_{n,m+k}\cdot C_{n,k,m}\,;k=1,2,\ldots18\,;\text{for } m<0$$

The average data subcarrier power is estimated for each equalized active partition *represented* by the outermost frequency reference subcarrier that bounds the partition.

$$P\_part_m = \frac{1}{18\cdot N}\sum_{n=0}^{N-1}\sum_{k=1}^{18}\left|v_{n,m-k}\right|^2\;;\;\text{for } m>0$$

$$P\_part_m = \frac{1}{18\cdot N}\sum_{n=0}^{N-1}\sum_{k=1}^{18}\left|v_{n,m+k}\right|^2\;;\;\text{for } m<0$$

The average partition power for each sideband and modulation type is calculated as follows:

1. QPSK

$$Pp\_part_{upper} = \left(\frac{1}{Mdat}\right) \bullet \sum_{part=0}^{Mdat-1} \left(P\_part_{Midx\,-\,19 \bullet part}\right)$$

$$Pp\_part_{lower} = \left(\frac{1}{Mdat}\right) \bullet \sum_{part=0}^{Mdat-1} \left(P\_part_{Midx+\,19 \bullet part}\right)$$

$$Ps\_part_{upper} = \left(\frac{1}{Mdat}\right) \bullet \sum_{part=0}^{Mdat-1} \left(P\_part_{Midx\,-\,19 \bullet part}\right)$$

$$Ps\_part_{lower} = \left(\frac{1}{Mdat}\right) \bullet \sum_{part=0}^{Mdat-1} \left(P\_part_{Midx\,+\,19 \bullet part}\right)$$

2. 16-QAM and 64-QAM

$$Pq\_part_{upper} = \left(\frac{1}{Mdat}\right) \bullet \sum_{part=0}^{Mdat-1} \left(P\_part_{Midx\,-\,19 \bullet part}\right)$$

$$Pq\_part_{lower} = \left(\frac{1}{Mdat}\right) \bullet \sum_{part=0}^{Mdat-1} \left(P\_part_{Midx\,+\,19 \bullet part}\right)$$

where *Mdat* is the number of active partitions of equal modulation type on one sideband and *Midx* is the starting index for the group of partitions (See Table 11-1 and Table 11-2).

Pp_part_{upper} is the average power of the partitions within the upper primary sideband that have a modulation type of QPSK.

Pp_part_{lower} is the average power of the partitions within the lower primary sideband that have a modulation type of QPSK.

Ps_part_{upper} is the average power of the partitions within the upper secondary sideband that have a modulation type of QPSK.

Ps_part_{lower} is the average power of the partitions within the lower secondary sideband that have a modulation type of QPSK.

Pq_part_{upper} is the average power of the partitions within the upper primary sideband that have a modulation type of 16-QAM or 64-QAM.

Pq_part_{lower} is the average power of the partitions within the lower primary sideband that have a modulation type of 16-QAM or 64-QAM.

Table 11-1: Active Partition Start Index (Midx) per Service Mode

Service Mode	Partition (*Midx*) Start index							
	Primary						Secondary	
	Lower Sideband			Upper Sideband			Lower Sideband	Upper Sideband
	QPSK	16-QAM	64-QAM	QPSK	16-QAM	64-QAM	QPSK	QPSK
MP1	−546	---	---	546	---	---	---	---
MP2	−546	---	---	546	---	---	---	---
MP3	−546	---	---	546	---	---	---	---
MP5	−546	---	---	546	---	---	---	---
MP6	−546	---	---	546	---	---	---	---
MP11	−546	---	---	546	---	---	---	---
MP1X	−546	-356	---	546	356	---	---	---
DSB1	---	−546	---	---	546	---	---	---
MP1XOV	---	-546	---	---	546	---	---	---
MP6OV	---	-546	---	---	546	---	---	---
DSB1OV	---	---	−546	---	---	546	---	---
MS5	---	---	---	---	---	---	-267	267

Table 11-2: Number of Active Data Partitions (Mdat) versus Service Mode and Modulation Type

Service Mode	*Mdat*							
	Primary						Secondary	
	Lower Sideband			Upper Sideband			Lower Sideband	Upper Sideband
	QPSK	16-QAM	64-QAM	QPSK	16-QAM	64-QAM	QPSK	QPSK
MP1	10	---	---	10	---	---	---	---
MP2	11	---	---	11	---	---	---	---
MP3	12	---	---	12	---	---	---	---
MP5	14	---	---	14	---	---	---	---
MP6	14	---	---	14	---	---	---	---
MP11	14	---	---	14	---	---	---	---
MP1X	10	4	---	10	4	---	---	---
DSB1	---	14	---	---	14	---	---	---
MP1XOV	---	14	---	---	14	---	---	---
MP6OV	---	14	---	---	14	---	---	---
DSB1OV	---	---	14	---	---	14	---	---
MS5	---	---	---	---	---	---	14	14

12 PAPR Reduction Algorithm Increases Subcarrier Power

The PAPR reduction (noise) causes a modest change in digital signal power (e.g., 0.5 dB), which must be adjusted back to the target power level. This post-PAPR reduction adjustment simply scales all subcarriers equally with a single gain factor by scaling the symbol in the time domain. Also, the power change of the reference subcarriers is generally different than the power change of the data subcarriers. Although we know that the present PAPR reduction algorithm requires only a small gain adjustment (e.g., 0.5 dB), the change in ratio between the reference and data subcarrier power levels should be limited.

The equalized power for each active Reference subcarrier is:

$$P\_ref_{\mathrm{m}} = \left(\frac{2}{N \cdot (smag_m)^2}\right) \cdot \sum_{n=0}^{N-1} \left|r_{n,m}\right|^2$$

The average Reference subcarrier power for the Primary upper and lower sidebands is:

$$Pp\_ref_{upper} = 1/\mathrm{Mref} \bullet \sum P\_ref_m \qquad for\ m >= 280$$

$$Pp\_ref_{lower} = 1/\mathrm{Mref} \bullet \sum P\_ref_m \qquad for\ m <= -280$$

where m refers to the active reference subcarriers and Mref refers to the total number of active reference subcarriers on a given Primary sideband.

The average Reference subcarrier power for the Secondary upper and lower sidebands is:

$$Ps\_ref_{upper} = 1/Mref \bullet \sum P\_ref_m \qquad for \qquad 1 <= m <= 267$$

$$Ps\_ref_{lower} = 1/Mref \bullet \sum P\_ref_m \qquad for\ -267 <= m <= -1$$

where m refers to the active reference subcarriers and Mref refers to the total number of active reference subcarriers on a given Secondary sideband.

The voltage ratio of the data subcarriers to the reference subcarrier for the Primary sidebands and modulation type is:

1. QPSK

$$Rp_{upper} = \sqrt{\frac{Pp\_part_{upper}}{Pp\_ref_{upper}}}$$

$$Rp_{lower} = \sqrt{\frac{Pp\_part_{lower}}{Pp\_ref_{lower}}}$$

2. 16-QAM and 64-QAM

$$Rq_{upper} = \sqrt{\frac{Pq\_part_{upper}}{Pp\_ref_{upper}}}$$

$$Rq_{lower} = \sqrt{\frac{Pq\_part_{lower}}{Pp\_ref_{lower}}}$$

The voltage ratio of the data subcarriers to the reference subcarrier for the Secondary sidebands is:

$$Rs_{upper} = \sqrt{\frac{Ps\_part_{upper}}{Ps\_ref_{upper}}}$$

$$Rs_{lower} = \sqrt{\frac{Ps\_part_{lower}}{Ps\_ref_{lower}}}$$

Equation 3: Computation of Data Subcarrier to Reference Subcarrier Power Ratios in dB

3(a) $RpdB_{upper} = 20 \bullet \log_{10}(Rp_{upper})$

3(b) $RpdB_{lower} = 20 \bullet \log_{10}(Rp_{lower})$

3(c) $RqdB_{upper} = 20 \bullet \log_{10}(Rq_{upper})$

3(d) $RqdB_{lower} = 20 \bullet \log_{10}(Rq_{lower})$

3(e) $RsdB_{upper} = 20 \bullet \log_{10}(Rs_{upper})$

3(f) $RsdB_{lower} = 20 \bullet \log_{10}(Rs_{lower})$

These ratios are related to the expected loss due to the PAPR reduction power increase, and subsequent power-level adjustment. The computed values of R are used in the data subcarrier MER measurements.

Note: In the Data MER expressions (below), R restores the proper level of the reference subcarriers for the computation and prevents the possibility of artificially lowering the reference subcarriers to improve MER results.

13 Data Subcarrier Partition MER

In this subsection, the noise metric computation for each partition is described, where *m* is the index of the active outermost Reference Subcarrier bounding the partition. Note that in this calculation, the reference subcarrier is not included as indicated with i starting with a value of 1. With QPSK modulation, the *R* value used should correspond to the appropriate Primary/Secondary *R* value (*Rp* or *Rs*) as calculated in Section 12, dependent upon the sideband on which the partition resides.

Note: The Primary sideband can have both Rp and Rq, while the Secondary sideband can have only Rs. Also note that there are no partitions corresponding to subcarrier indices -280, -1, 1, and 280.

1. QPSK modulation

$$Vndat_m = \sqrt{\frac{1}{N\cdot 18}\cdot\sum_{k=1}^{18}\sum_{n=0}^{N-1}\left[\left(\max\left\{0, R_{upper} - \left|\mathrm{Re}\left\{v_{n,m-k}\right\}\right|\right\}\right)^2 + \left(\max\left\{0, R_{upper} - \left|\mathrm{Im}\left\{v_{n,m-k}\right\}\right|\right\}\right)^2\right]}$$

$\text{for}\, m > 0$

$$Vndat_m = \sqrt{\frac{1}{N\cdot 18}\cdot\sum_{k=1}^{18}\sum_{n=0}^{N-1}\left[\left(\max\left\{0, R_{lower} - \left|\mathrm{Re}\left\{v_{n,m+k}\right\}\right|\right\}\right)^2 + \left(\max\left\{0, R_{lower} - \left|\mathrm{Im}\left\{v_{n,m+k}\right\}\right|\right\}\right)^2\right]}$$

$\text{for}\; m < 0$

2. 16-QAM modulation

$$Vndat_m = \sqrt{\frac{1}{N\cdot 18}\cdot\sum_{k=1}^{18}\sum_{n=0}^{N-1}\left[\left(\max\{L16(\mathrm{Re}\{v_{n,m-k}\}), Rq_{upper}\bullet W16(\mathrm{Re}\{v_{n,m-k}\})-|\mathrm{Re}\{v_{n,m-k}\}|\}\right)^2 + \left(\max\{L16(\mathrm{Im}\{v_{n,m-k}\}), Rq_{upper}\bullet W16(\mathrm{Im}\{v_{n,m-k}\})-|\mathrm{Im}\{v_{n,m-k}\}|\}\right)^2\right]}$$

$\text{for}\; m > 0$

$$Vndat_m = \sqrt{\frac{1}{N\cdot 18}\cdot\sum_{k=1}^{18}\sum_{n=0}^{N-1}\left[\left(\max\{L16(\mathrm{Re}\{v_{n,m+k}\}), Rq_{lower}\bullet W16(\mathrm{Re}\{v_{n,m+k}\})-|\mathrm{Re}\{v_{n,m+k}\}|\}\right)^2 + \left(\max\{L16(\mathrm{Im}\{v_{n,m+k}\}), Rq_{lower}\bullet W16(\mathrm{Im}\{v_{n,m+k}\})-|\mathrm{Im}\{v_{n,m+k}\}|\}\right)^2\right]}$$

$\text{for}\; m < 0$

Where:

$$W16(x) = \begin{cases} 3/\sqrt{5} & \text{for}\quad 2/\sqrt{5} < |x| \\ 1/\sqrt{5} & \text{for}\quad 2/\sqrt{5} >= |x| \end{cases}$$

Function W16(x) retrieves a weighting value that places Rp at the center of a 16-QAM constellation cell, represented by the data modulating a data subcarrier.

and:

$$L16(x) = \begin{cases} 0 & \text{for } 2/\sqrt{5} < |x| \\ -\infty & \text{for } 2/\sqrt{5} >= |x| \end{cases}$$

Function L16(x) retrieves a value that allows any noise to contribute to the noise power estimate.

3. 64-QAM modulation

$$Vndat_m = \sqrt{\frac{1}{N \cdot 18} \cdot \sum_{k=1}^{18} \sum_{n=0}^{N-1} \left[(\max\{L64(\mathrm{Re}\{v_{n,m-k}\}), Rq_{upper} \bullet W64(\mathrm{Re}\{v_{n,m-k}\}) - |\mathrm{Re}\{v_{n,m-k}\}|\})^2 + (\max\{L64(\mathrm{Im}\{v_{n,m-k}\}), Rq_{upper} \bullet W64(\mathrm{Im}\{v_{n,m-k}\}) - |\mathrm{Im}\{v_{n,m-k}\}|\})^2 \right]}$$

for $m > 0$

$$Vndat_m = \sqrt{\frac{1}{N \cdot 18} \cdot \sum_{k=1}^{18} \sum_{n=0}^{N-1} \left[(\max\{L64(\mathrm{Re}\{v_{n,m+k}\}), Rq_{lower} \bullet W64(\mathrm{Re}\{v_{n,m+k}\}) - |\mathrm{Re}\{v_{n,m+k}\}|\})^2 + (\max\{L64(\mathrm{Im}\{v_{n,m+k}\}), Rq_{lower} \bullet W64(\mathrm{Im}\{v_{n,m+k}\}) - |\mathrm{Im}\{v_{n,m+k}\}|\})^2 \right]}$$

for $m < 0$

Where:

$$W64(x) = \begin{cases} 7/\sqrt{21} & \text{for } 6/\sqrt{21} < |x| \\ 5/\sqrt{21} & \text{for } 4/\sqrt{21} < |x| <= 6/\sqrt{21} \\ 3/\sqrt{21} & \text{for } 2/\sqrt{21} < |x| <= 4/\sqrt{21} \\ 1/\sqrt{21} & \text{for } |x| <= 2/\sqrt{21} \end{cases}$$

Function W64(x) retrieves a weighting value that places Rp at the center of a 64-QAM constellation cell, represented by the data modulating a data subcarrier.

and:

$$L64(x) = \begin{cases} 0 & \text{for } 6/\sqrt{21} < |x| \\ -\infty & \text{for } 6/\sqrt{21} >= |x| \end{cases}$$

Function L64(x) retrieves a value that allows any noise to contribute to the noise power estimate.

Once $Vndat_m$ has been calculated, the MER can be obtained from the following equation:

Equation 4: MER per 18-Data-Subcarrier Partition

$$MERdat_m = -20 \cdot \log_{10}\{\, Vndat_m \}$$

This computes the MER for a single upper-sideband or lower-sideband partition. Equation 5 shows how to generate a composite MER value for an entire sideband.

Mdat is the number of active partitions on one sideband (Primary/Secondary upper or lower) of like modulation (see Table 11-2).

Midx is the starting reference subcarrier number of the outermost active partition (see Table 11-1).

MERdatp is for Primary sideband partitions with QPSK modulation.

MERdatq is for Primary sideband partitions with 16-QAM or 64-QAM modulation.

MERdats is for Secondary sideband partitions with QPSK modulation.

Equation 5: Composite MER Measurement for Upper-Sideband and Lower-Sideband Data Subcarriers with like Modulation Types

5(a) $$MERdatp_{upper} = 10 \cdot log_{10}\left(\left(\frac{1}{Mdat}\right) \cdot \sum_{part=0}^{Mdat-1} 10^{\left(\frac{(MERdat_{Midx-(19 \cdot part)})}{10}\right)} \right)$$

5(b) $$MERdatp_{lower} = 10 \cdot log_{10}\left(\left(\frac{1}{Mdat}\right) \cdot \sum_{part=0}^{Mdat-1} 10^{\left(\frac{(MERdat_{Midx+(19 \cdot part)})}{10}\right)} \right)$$

5(c) $$MERdatq_{upper} = 10 \cdot log_{10}\left(\left(\frac{1}{Mdat}\right) \cdot \sum_{part=0}^{Mdat-1} 10^{\left(\frac{(MERdat_{Midx-(19 \cdot part)})}{10}\right)} \right)$$

5(d) $$MERdatq_{lower} = 10 \cdot log_{10}\left(\left(\frac{1}{Mdat}\right) \cdot \sum_{part=0}^{Mdat-1} 10^{\left(\frac{(MERdat_{Midx+(19 \cdot part)})}{10}\right)} \right)$$

5(e) $$MERdats_{upper} = 10 \cdot log_{10}\left(\left(\frac{1}{Mdat}\right) \cdot \sum_{part=0}^{Mdat-1} 10^{\left(\frac{(MERdat_{Midx-(19 \cdot part)})}{10}\right)}\right)$$

5(f) $$MERdats_{lower} = 10 \cdot log_{10}\left(\left(\frac{1}{Mdat}\right) \cdot \sum_{part=0}^{Mdat-1} 10^{\left(\frac{(MERdat_{Midx+(19 \cdot part)})}{10}\right)}\right)$$

When computing composite MERs for an entire sideband, only like-modulated partition MERs can be averaged together. This can lead to multiple sideband MER values. For example, service mode MP1X will have an upper-sideband MER for QPSK modulation with *Mdat* = 10 and *Midx* = 546. It will also have an upper-sideband MER for 16-QAM modulation with *Mdat* = 4 and *Midx* = 356.

The above equations may be used to assess whether the digital transmission system is in compliance with the requirements stated in Subsection 4.8 of Reference [7].

The MER is inclusive of any distortion caused by the PAPR reduction algorithm as well as any local variations in amplitude and group delay not removed by the equalizer described in Section 11. Thus, it should be indicative of actual receiver performance.

The intent of the specifications proposed in this document is to utilize MER to evaluate the performance of the transmission system equipment as measured at the RF output connection point to the antenna system. However, it must be noted that the same MER computations could be accomplished by a monitor receiver located at the broadcaster installation in order to evaluate the effects of the antenna system, as well as any interference to the digital subcarriers caused by an overmodulated or otherwise distorted analog FM host signal. Xperi will not impose specific requirements on this. But they may offer voluntary recommendations in the future once the MER specifications have been fully characterized. For example, the broadcaster could utilize a MER monitor receiver to signal an alarm indicating a fault condition that may be compromising receiver performance in the field.

Appendix A Frequency and Symbol Timing Acquisition from FM IBOC MER Measurement

This section provides a suggestion for performing synchronization of the carrier frequency and symbol timing.

The sample rate, fs=744,187.5 Hz. Symbol timing is less sensitive to sample rate frequency errors than is the carrier phase. So the sample rate frequency tolerances discussed in Section 6 apply here as well.

If the analog FM signal is present, it must first be removed in order for the techniques described here to work properly.

In order to process *N* "complete" symbols, each consisting of *K*=2160 complex signal samples, an additional symbol is buffered to accommodate the possibility of partial symbols at the boundaries (the minimum number of required additional samples is actually one sample less than a whole symbol). The total number of input complex signal samples is then $(N+1)\cdot K$.

The sequence of signal samples x_n , $n=0,\ldots,[(N+1)\cdot K]-1$, is complex-conjugate multiplied by the sequence delayed by 2048 samples (i.e., x_{n+2048}). The products are stored in a single-symbol, *K*-sample complex buffer by folding modulo *K*. Notice that only N symbols are processed (not N+1) to allow for the delay.

$$y_{\mathrm{mod}(n,K)} = y_{\mathrm{mod}(n,K)} + x_n \cdot x^*_{n+2048} \quad ; \text{ for } n=0,\ldots,\mathrm{N}\cdot K-1$$

where * indicates complex conjugate, and *y* is previously initialized to zero because of the recursive operation. Next the vector *y* is match-filtered, using cyclic convolution, for the expected cyclic prefix autocorrelation pulse shape *h*, to produce vector *v*.

$$v_k = \sum_{m=1}^{111} h_m \cdot y_{\mathrm{mod}(k+m,K)} \quad ; \; k=0,\ldots,K-1$$

$$\text{where } h_m = \sin\left(\frac{m\cdot\pi}{112}\right) \quad ; \; m=0,\ldots 111$$

The sample offset error is simply computed as the index of the peak magnitude sample of the vector *v*. This corresponds to the starting (zero index) sample of the first complete symbol. Symbol synchronization to the nearest whole sample is sufficient, since the subsequent equalization process will remove the linear phase distortion resulting from residual symbol timing error.

$$samperr = \text{ index } k \;\; ; \;\; \text{corresponding to } \max_k |v_k|$$

The estimated frequency error computed in Hz is

$$freqerr = \frac{-fs}{2\cdot\pi\cdot 2048}\cdot\arctan\left(\frac{\mathrm{Im}\{v_{samperr}\}}{\mathrm{Re}\{v_{samperr}\}}\right)$$

The sample and frequency error estimates are then used to adjust the original sequence, and output exactly N synchronized symbols.

$$xadj_n = x_{(n+samperr)}\cdot\exp\{-j\cdot 2\cdot\pi\cdot freqerr\cdot n/fs\} \quad ; \text{ for } n=0,\ldots,K\cdot\mathrm{N}-1$$

OFDM Demodulation of the Block of N Symbols

Demodulation of the block of N OFDM symbols is next. Define the root-raised cosine Nyquist pulse shape window function for k=0,…,K-1, and K=2160.

$$\mathrm{win}_k = \begin{cases} \sin\left(\dfrac{k \cdot \pi}{224}\right) & ;\ \textit{if}\ \ k < 112 \\ \sin\left(\dfrac{(K-k)\cdot \pi}{224}\right) & ;\ \textit{if}\ \ k > 2048 \\ 1 & ;\ \ \textit{otherwise} \end{cases}$$

The demodulation of a signal vector consisting of a block of N synchronized symbols can be computed as follows. First initialize the matrix *sig* for the appropriate size (2048 rows by N columns), and zero-fill to accommodate the initial condition of the recursive computation.

$$sig_{2047,N-1} = 0$$

Arrange the signal column vector into a matrix of N symbol column vectors. Then window and fold the ending cyclic suffix onto the prefix.

$$sig_{\mathrm{mod}\left[n-floor\left(\frac{n}{K}\right)\cdot K,2048\right],\, floor\left(\frac{n}{K}\right)} = sig_{\mathrm{mod}\left[n-floor\left(\frac{n}{K}\right)\cdot K,2048\right],\, floor\left(\frac{n}{K}\right)} + xadj_n \cdot win_{\mathrm{mod}[n,K]}\ ;$$
$$n = 0,\ldots,K\cdot N-1\ \text{(i.e., } N \text{ symbols)}$$

Demodulate each column (OFDM symbol) of *sig* using complex FFT, and appropriate scaling such that constellation points are nominally (±1, ±1), with possible phase rotation.

$$OFDM\_VECTOR(\text{column } m) = CFFT\left(sig(\text{column } m)\right)\cdot\sqrt{\frac{256\cdot 2\cdot 191}{135}}\quad ;\ m\text{=0,… N-1}$$

A symbol timing error (*samperr*) results in a phase rotation of the constellation across subcarriers (k). There is no phase rotation over time (not affected by m). The phase rotation is resolved by the equalizer.

A frequency error (*freqerr*) results in a phase rotation of the constellation for any one subcarrier over time (m). Even one Hz error causes more than one full rotation over 512 symbols. However, all subcarriers are in phase sync. This frequency estimate algorithm is very accurate, and should result in no significant phase rotation. The fixed phase offset is arbitrary, and is resolved by the equalizer (phase just happens to be zero in this simulation).

MathCad Example

Generate one more symbol than the required target to accommodate partial symbols at the boundaries:

Number of symbols $N := 64$ $\quad$ $samperr := 0$ $\quad$ $freqerr := 0$ $\quad$ $x := FM\_dig\_sig(N+1, samperr, freqerr)$

Samples per symbol K: $K := 2160$ $\quad$ $k := 0..K-1$

Process samples allowing for delay of 2048 samples (~1 symbol): $n := 0..N\cdot K - K - 1$

Complex multiply delayed samples and fold symbols: $y_{K-1} := 0$ $\quad$ $y_{mod(n,K)} := y_{mod(n,K)} + x_n \cdot \overline{x_{n+2048}}$

Create matched filter for autocorrelation pulse shape: $m := 0..111$ $\quad$ $h_m := \sin\left(\frac{m\cdot\pi}{112}\right)$

Matched filter for the resulting autocorrelation vector: $v_k := \sum_{m=1}^{111} \left(h_m \cdot y_{mod(k+m,K)}\right)$

Compute the sample and frequency errors from v:

$$
sampfreqerr(v) := \left|
\begin{array}{l}
\text{"find peak and its location"} \\
samperr \leftarrow 0 \\
pk \leftarrow |v_0| \\
\text{for } k \in 1..K-1 \\
\quad \text{if } |v_k| > pk \\
\qquad \left| \begin{array}{l} samperr \leftarrow k \\ pk \leftarrow |v_k| \end{array} \right. \\
\begin{pmatrix} samperr \\ freqerr \leftarrow -\frac{fs}{2\cdot\pi\cdot 2048}\cdot \operatorname{atan}\left(\frac{\operatorname{Im}(v_{samperr})}{\operatorname{Re}(v_{samperr})}\right) \end{pmatrix}
\end{array}
\right.
$$

$$\begin{pmatrix} samperr \\ freqerr \end{pmatrix} := sampfreqerr(v)$$

$$\begin{pmatrix} samperr \\ freqerr \end{pmatrix} = \begin{pmatrix} 0 \\ 0 \end{pmatrix}$$

Adjust signal x for sample and freq errors, 64 symbols: $n := 0..N\cdot 2160 - 1$ $\quad$ $xadj_n := x_{n+samperr} \cdot e^{-j\cdot 2\cdot\pi\cdot freqerr\cdot\frac{n}{fs}}$

$var(x) = 1$

Figure A-1: Symbol Synchronization and Frequency Acquisition

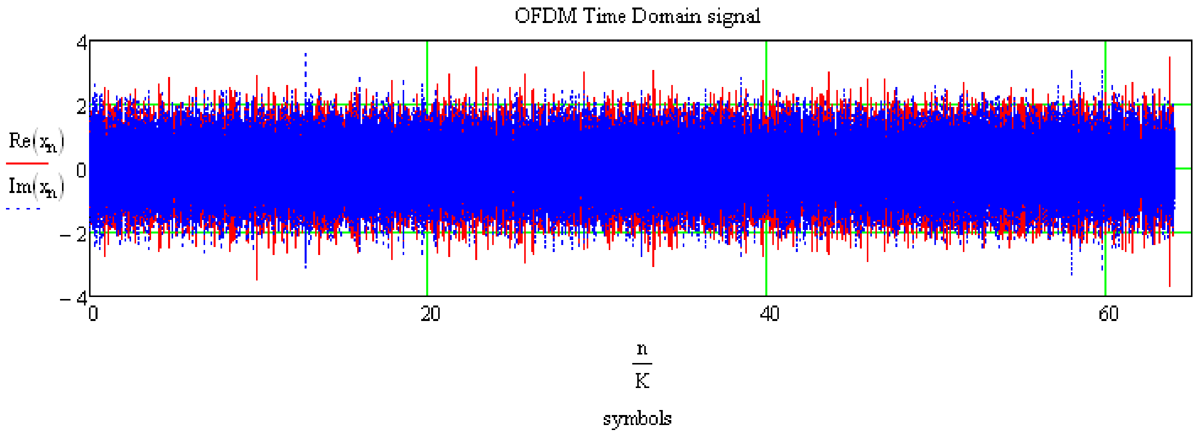


Figure A-2: OFDM Time Domain Signal

SIG := CFFT(x)

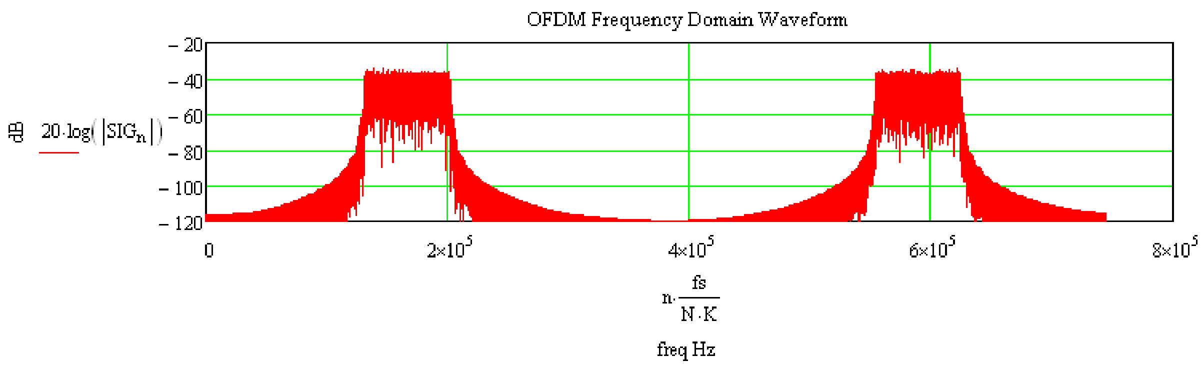


Figure A-3: OFDM Frequency Domain Waveform

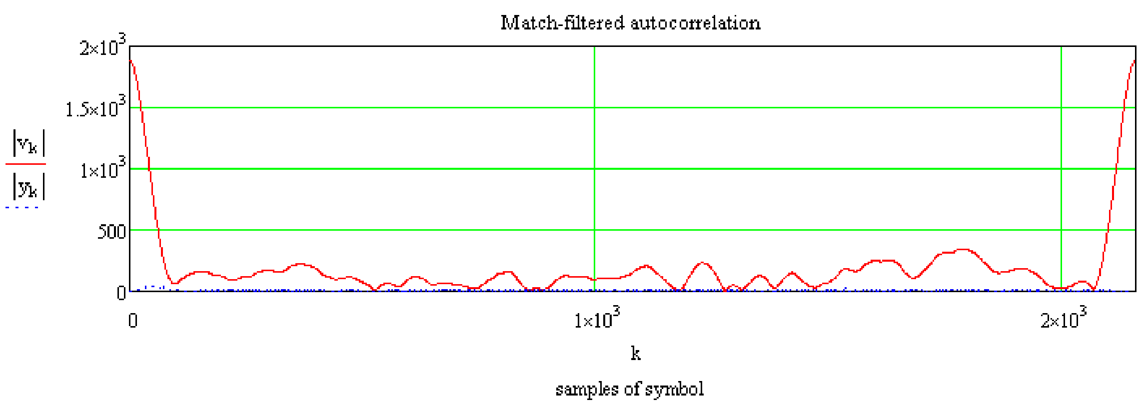


Figure A-4: Match-Filtered Autocorrelation

Demodulate synchronized OFDM signal:

Apply window to matrix:

$$\text{win} := \left| \begin{array}{l} \text{"Generate 2160-sample window for pulse shape"} \\ Nw \leftarrow 2048 \\ TAPER \leftarrow 112 \\ \text{for } nt \in 0..Nw + TAPER - 1 \\ \quad w_{nt} \leftarrow \begin{cases} \sin\left(nt \cdot \dfrac{\pi}{2 \cdot TAPER}\right) & \text{if } nt < TAPER \\ \sin\left[(Nw + TAPER - nt) \cdot \dfrac{\pi}{2 \cdot TAPER}\right] & \text{if } nt > Nw \\ 1 & \text{otherwise} \end{cases} \\ w \end{array} \right.$$

Arrange signal vector xadj into matrix of symbol vector columns: Fold ending cyclic suffix onto prefix:

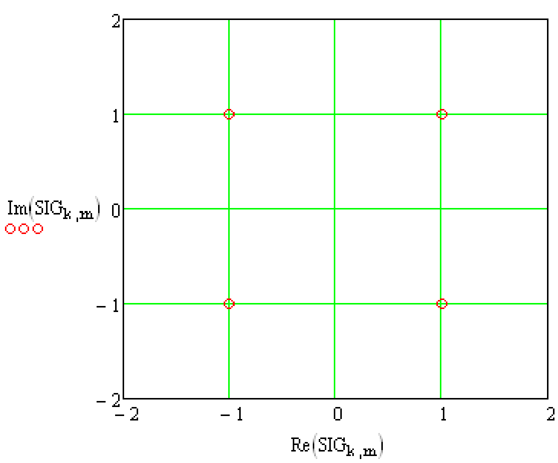
$$sig_{2047,63} := 0 \qquad sig_{\text{mod}\left(n-\text{floor}\left(\frac{n}{2160}\right)\cdot 2160,2048\right),\text{floor}\left(\frac{n}{K}\right)} := sig_{\text{mod}\left(n-\text{floor}\left(\frac{n}{2160}\right)\cdot 2160,2048\right),\text{floor}\left(\frac{n}{K}\right)} + xadj_n \cdot win_{\text{mod}(n,K)}$$

$k := 1502$

$m := 0..63$

$k := 1692$

$$SIG^{\langle m \rangle} := \text{CFFT}\left(sig^{\langle m \rangle}\right) \cdot \sqrt{\frac{256 \cdot 2 \cdot 191}{135}} \qquad k := 546 \qquad k := 356..546 \qquad \begin{pmatrix} samperr \\ freqerr \end{pmatrix} = \begin{pmatrix} 0 \\ 0 \end{pmatrix}$$



A symbol timing error (samperr) results in a phase rotation of the constellation across subcarriers (k). There is no phase rotation over time (not affected by m). The phase rotation is resolved by the equalizer.

A frequency error (freqerr) results in a phase rotation of the constellation for any one subcarrier over time (m). Even one Hz error causes more than 1 full rotation over 512 symbols. However, all subcarriers are in phase sync. This frequency estimate algorithm is very accurate, and should result in no significant phase rotation. The fixed phase offset is arbitrary, and is resolved by the equalizer (phase just happens to be zero in this simulation).

A sample rate error results in a phase rotation of the constellation for any one subcarrier over time (m). For example, if the sample rate error causes a drift of 1 sample over the signal time span, then this would result in a phase shift of 360*fc/fs degrees, or about 97 degrees for a subcarrier at 200 kHz from center frequency.

Figure A-5: Demodulated OFDM Symbol Constellation

Appendix B Computed MER Values and Measured Receiver Performance Using iBiquity Reference Test Data for Initial Version of PAPR Reduction Algorithm

Table B-1 shows the MER values computed from the reference data files supplied by iBiquity Digital. The block size (N) is 512.

Figure B-1 plots the P1 Channel BER that is tabulated in Table B-1, along with plots of the PIDS block error ratio (BLKER).

Table B-1: Computed MER Values and Measured Receiver Performance

Test File Name	Service Mode	PAPR Reduction On/Off	Cd/No dB-Hz	MER(ref)$_{Avg}$	MER(ref)$_{Worst Case}$ / Subcarrier Index	MER(dat)$_{Avg}$	MER(dat)$_{Worst Case}$ / Partition Index	P1 Channel BER
MP1_PAROff_52dB	MP1	Off	52	1.5	1.0	4.8	4.7	7.80E-03
MP1_PAROff_54dB	MP1	Off	54	3.1	2.8	5.2	5.0	3.30E-05
MP1_PAROff_56dB	MP1	Off	56	5.0	4.7	6.0	5.8	1.10E-07
MP1_PAROff_58dB	MP1	Off	58	6.9	6.6	7.2	7.0	0
MP1_PAROff_60dB	MP1	Off	60	8.9	8.5	8.9	8.7	0
MP1_PAROff_62dB	MP1	Off	62	10.9	10.5	10.8	10.6	0
MP1_PAROff_64dB	MP1	Off	64	12.8	12.5	12.8	12.6	0
MP1_PAROff_66dB	MP1	Off	66	14.8	14.5	14.8	14.6	0
MP1_PAROff_68dB	MP1	Off	68	16.8	16.5	16.8	16.5	0
mp1_PARoff_noAnalog	MP1	Off	No noise	88.7	88.2	88.6	88.1	0
MP1_PAROn_52dB	MP1	On	52	1.0	0.54	4.6	4.4	1.10E-02
MP1_PAROn_54dB	MP1	On	54	2.6	2.2	4.8	4.7	6.80E-05
MP1_PAROn_56dB	MP1	On	56	4.4	4.0	5.5	5.3	5.70E-08
MP1_PAROn_58dB	MP1	On	58	6.3	5.8	6.6	6.4	0
MP1_PAROn_60dB	MP1	On	60	8.1	7.6	8.0	7.8	0
MP1_PAROn_62dB	MP1	On	62	10.0	9.5	9.6	9.3	0
MP1_PAROn_64dB	MP1	On	64	11.8	11.3	11.2	10.9	0
MP1_PAROn_66dB	MP1	On	66	13.6	13.1	12.6	12.3	0
MP1_PAROn_68dB	MP1	On	68	15.2	14.7	13.9	13.6	0
mp1_PARon_noAnalog	MP1	On	No noise	21.6	21.0	18.0	17.5	0

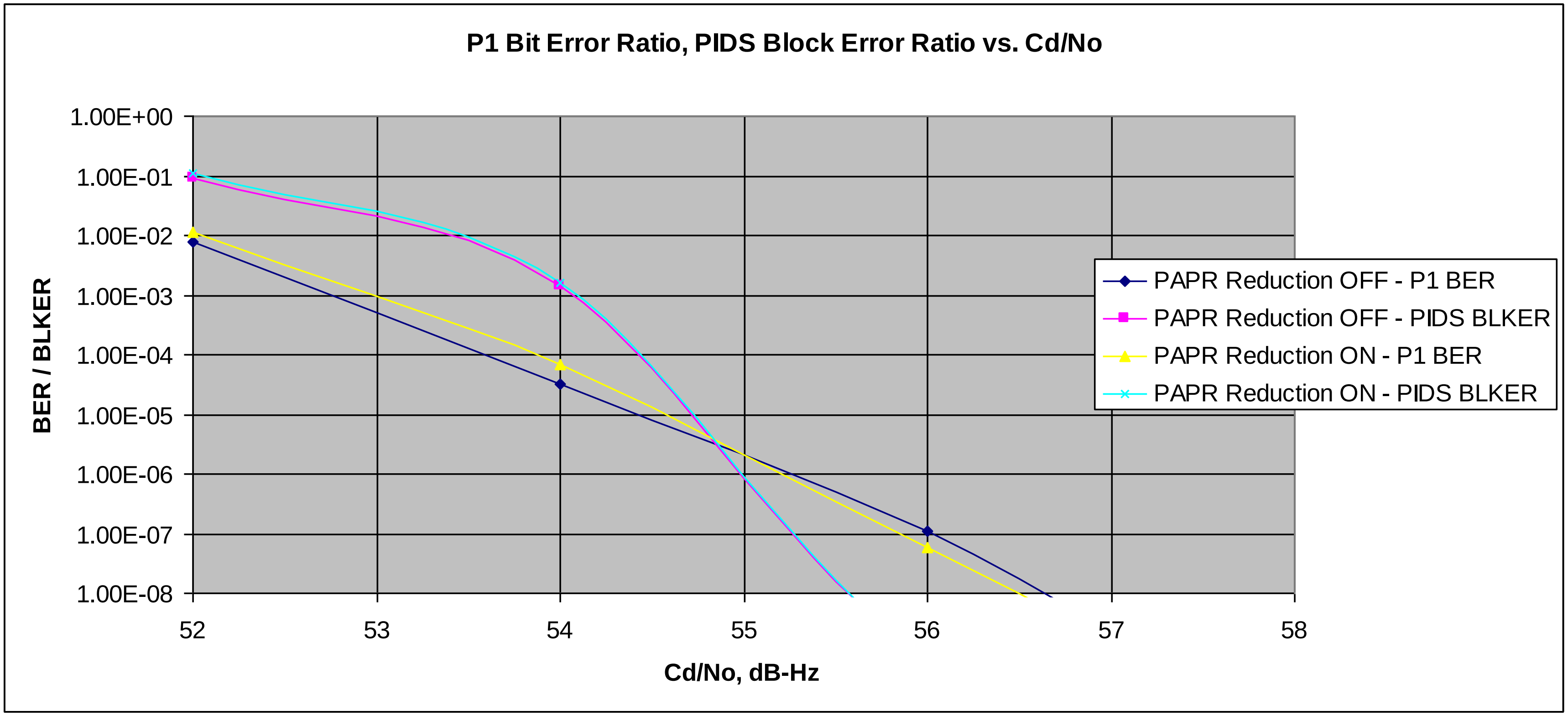


Figure B-1: P1 Bit Error Ratio and PIDS Block Error Ratio versus Cd/No

Discussion

From the noise vectors analyzed here, it is apparent that as white Gaussian noise is added to the signal, the noise quickly dominates over the PAPR-reduction-induced constellation noise before bit errors become apparent. This statement is supported by comparing the MER numbers for the PAPR-reduction-enabled and PAPR-reduction-disabled cases versus the signal without added noise. The difference in MER numbers quickly diminishes with added noise, indicating that the introduced noise is dominating.

Bit errors are only detected when Cd/No=56 dB-Hz, where the difference in MER between the PAPR-reduction-enabled and PAPR-reduction-disabled cases is only about 0.5 dB.

The argument could be made that as long as the PAPR-reduction-induced constellation noise, as measured by the IBOC quality metric, is sufficiently better than the MER of the PAPR-reduction-disabled signal with added noise, then the impact of the PAPR-reduction-induced noise should be minimal to the received signal in an AWGN channel. So in the case where Cd/No=56 dB-Hz, the PAPR-reduction-disabled signal shows bit errors at a MER of 6.0 dB for data carriers and 5.0 dB for reference carriers. The PAPR-reduction-induced MER of 18.0 dB for data carriers and 21.6 dB for reference carriers provides a rather large margin before the system breaks down.

It is, however, prudent to add significant margin to account for multipath and mobile channel environments. The following analysis provides a method to derive a suitable MER specification point with a reasonable margin taken into account.

Consider a receiver that, due to channel noise, is operating approximately 2 dB above the blending point. According to the BER data, this is at Cd/No=56 dB-Hz. This is the point where the BER is around 10^{-7}; or almost error free. Assume for this analysis that the transmission system is perfect; that is, equivalent to the “PAPR Reduction Off – No Added Noise” reference file (mp1_PARoff_noAnalog).

It is recommended that, when a realistic transmission system is introduced into the above scenario, the total Cd/No of the link shall not be degraded by more than 0.5 dB; that is, Cd/No(total) ≥ 55.5 dB-Hz.

Cd/No(total) = Combined Cd/No of channel and transmitter.

This is computed by combining the noise from the two sources: No(total) = No(channel) + No(transmitter).

This computation must be a linear addition; so the No values must be converted from dB to linear.

For 0.5-dB degradation:

Cd/No(total) = 55.5 dB-Hz

$55.5 = -10*\log_{10}\{10^{-56/10} + 10^{-Cd/No(transmitter)}\}$

Solving the above equation for Cd/No(transmitter), the following result is obtained:

Cd/No(transmitter) = 65 dB-Hz

Referring to Table B-1, for the PAPR Reduction Off case, Cd/No=65 dB-Hz produces an average MER value of 13.8 dB for both the data and reference subcarriers. This can be obtained through interpolation of the MER cases for Cd/No= 64 dB-Hz and Cd/No=66 dB-Hz.

Rounding this average MER value to the nearest dB produces a specification of 14 dB. Allowing for a 3-dB margin for the worst case, the specification would be 11 dB. See Reference [7].

Appendix C Computed MER Values and Measured Receiver Performance Using Xperi Reference Test Data and PAPR Reduction Version 3

The following information describes how the MER specifications in Reference [7] were determined for each advanced FM service mode. Version 3 of the PAPR reduction algorithm was used to generate a series of target MER values. The PAPR of the transmitted signal and the bit error rate (BER) degradation of the received signal at the digital point of failure was measured for each target MER value. The MER specification for each service mode was then selected by analyzing the PAPR reduction and BER degradation across the range of target MERs.

C.1 Effects of PAPR Reduction Parameters Variation

Version 3 of the PAPR Reduction algorithm has three parameters that affect peak-to-average-power ratio reduction, MER, and received BER performance. In general, improving PAPR reduction results in noisier signal constellations that degrade the MER of the transmitted signal and the BER of the received signal.

The three PAPR reduction parameters are:

- IT: the number of PAPR Reduction algorithm iterations
- PD: The predistortion ratio, which adjusts the allowable variation (due to PAPR reduction) of the transmitted constellation points from an undistorted constellation point. A value of one allows no PAPR reduction distortion, while a value of zero would allow distortion all the way to the decision region boundaries.
- CBW: The convergence bias weight, which reduces the effects of PAPR reduction on signals with lower sideband power levels. A weight of 100 results in no distortion to the constellation points of the lower-powered sideband, while a weight of zero allows maximum distortion.

The general effects on MER, BER, and PAPR reduction of varying these parameters are shown in Table C-1. Example constellations with Version 3 PAPR Reduction are illustrated in Figure C-1 for QPSK and 16-QAM modulation.

Table C-1: Parametric Variation Effects on MER, BER, and PAPR Reduction

PAPR Reduction Parameter Variation	Improvement/Degradation		
	MER	**BER**	**PAPR Reduction**
IT Increase	Degrade	Degrade	Improve
PD Increase	Improve	Improve	Degrade
CBW Increase	Improve	Improve	Degrade

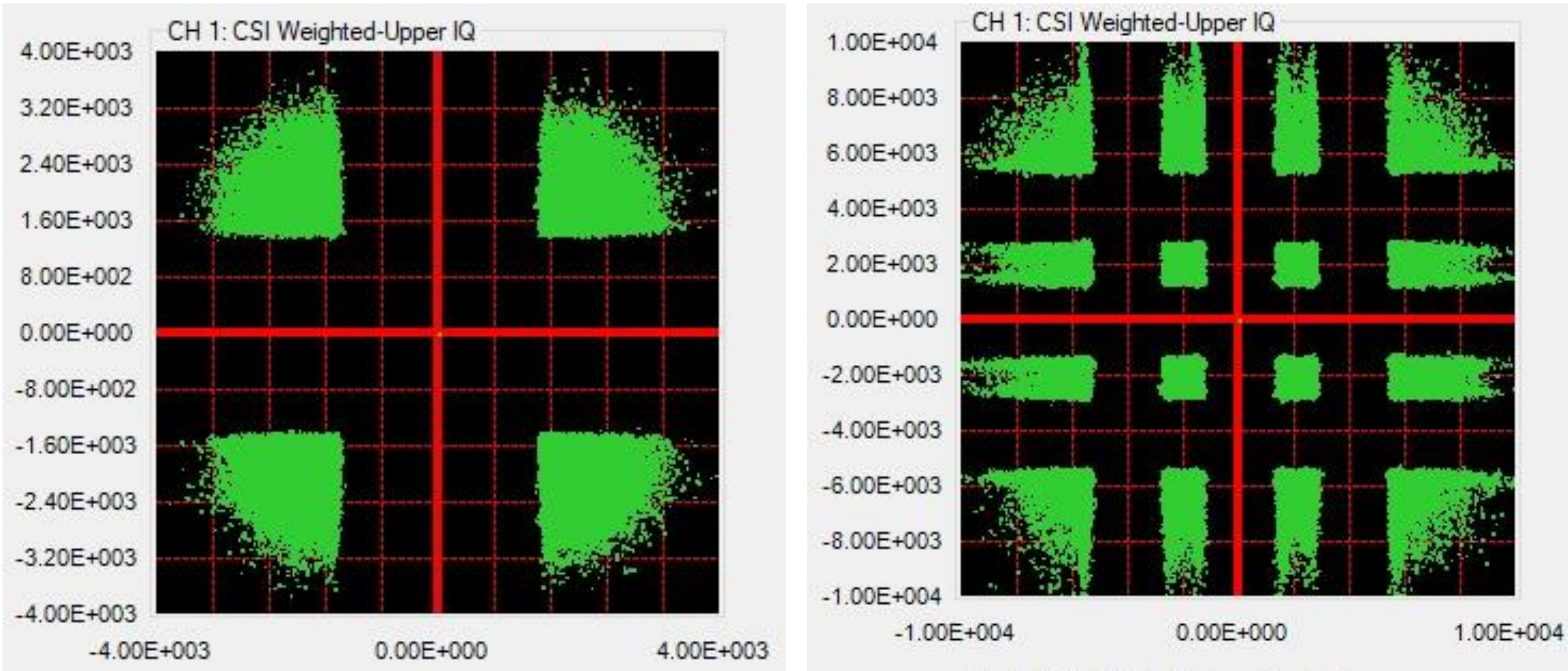


Figure C-1: Example Constellation Degradation due to PAPR Reduction for QPSK and 16-QAM

C.2 Process of Setting MER Specifications

The following process is used to set the MER specification for a given service mode. This is accomplished by adjusting Version 3 PAPR Reduction parameters for each target MER value, and then measuring the associated PAPR reduction and BER degradation. The MER with the lowest PAPR reduction resulting in acceptable BER degradation is then selected as the specification.

This process for each service mode is described in detail below:

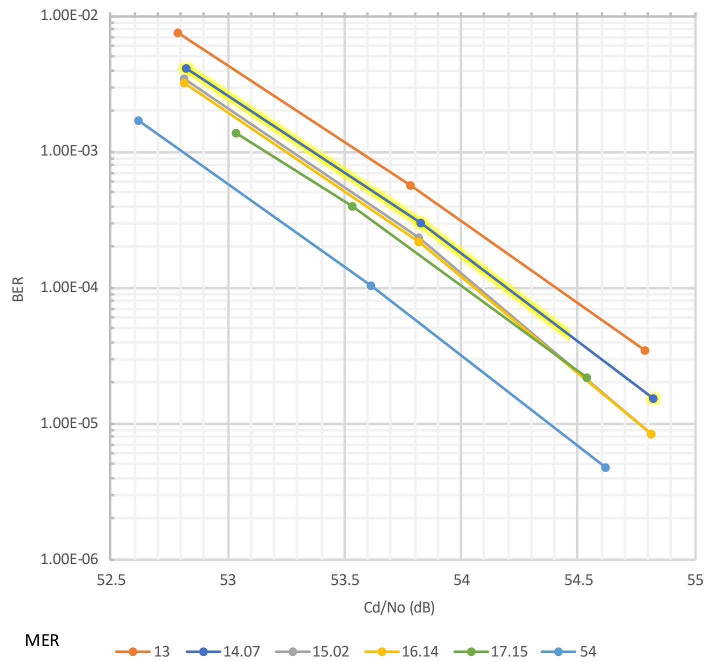
1. Select a service mode and logical channel
2. Select a target MER value
3. Adjust IT, PD, and CBW (in that order) to achieve the target MER value
4. Run Step 3 a second time
5. Run tests or simulations to measure receiver BER performance
6. Record the PAPR reduction value and BER degradation near the digital audio point of failure (5e-5)
7. Repeat steps (1) – (6) for the next targeted MER values
8. Once all targeted MER values have been tested, select a MER specification that provides sufficient PAPR reduction while maintaining acceptable BER degradation

C.3 MER Specification Test Results for Advanced FM Service Modes

The simulation and test results used to set the MER specifications defined in Reference [7], using the process defined in Subsection C.2, are provided below for advanced FM service modes. All BER simulations and tests were performed in an additive white Gaussian noise (AWGN) environment. Note that the selected MER specification for MP1 matches that determined using the initial version of the PAPR Reduction algorithm.

In Figure C-2 through Figure C-11, the chart on the left indicates BER performance versus Cd/No for each target MER value, where Cd is the total power of the digital signal, and No is the noise power in a 1- Hz bandwidth. The chart on the right indicates the PAPR reduction in dB versus MER for each target MER value. The specification for a given service mode is selected by choosing the target MER that has the lowest PAPR reduction with acceptable BER degradation. The selected BER curve and PAPR reduction value are highlighted in yellow for each service mode.

The MER specifications for advanced FM service modes are summarized in Table C-1.



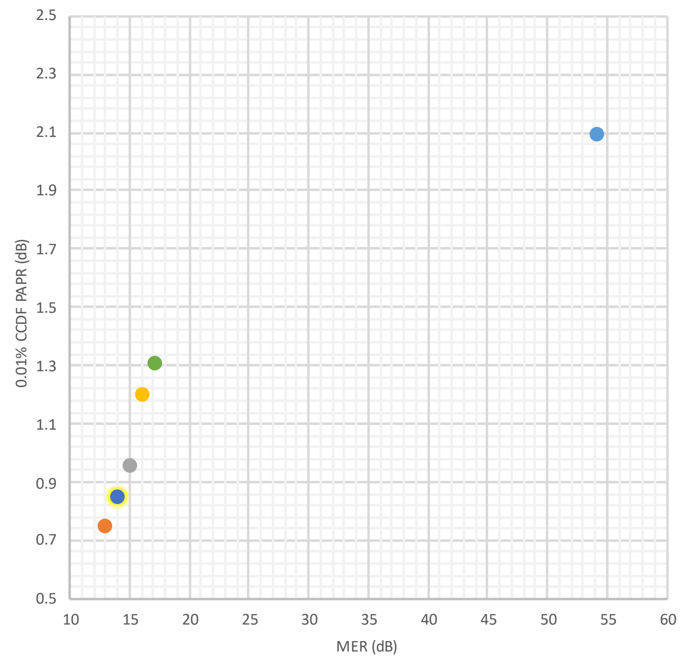
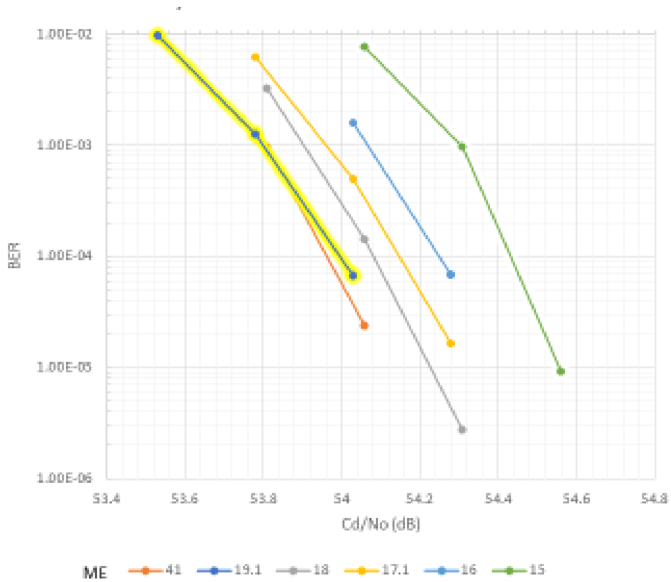


Figure C-2: BER Curves and PAPR Reduction versus MER for Hybrid QPSK MP1X P1



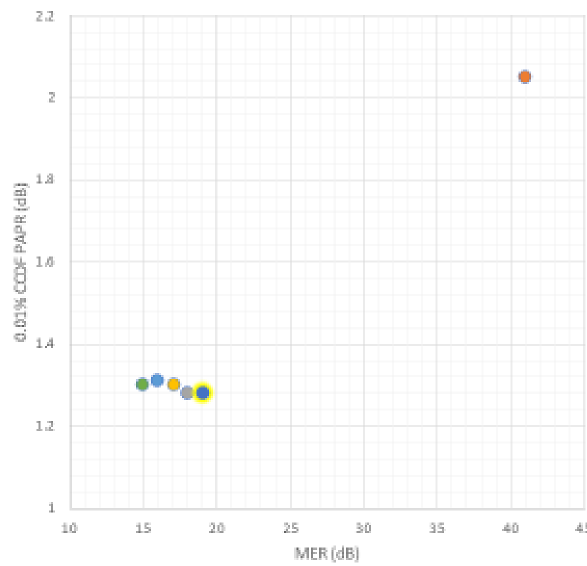


Figure C-3: BER Curves and PAPR Reduction versus MER for Hybrid 16-QAM MP1X P4

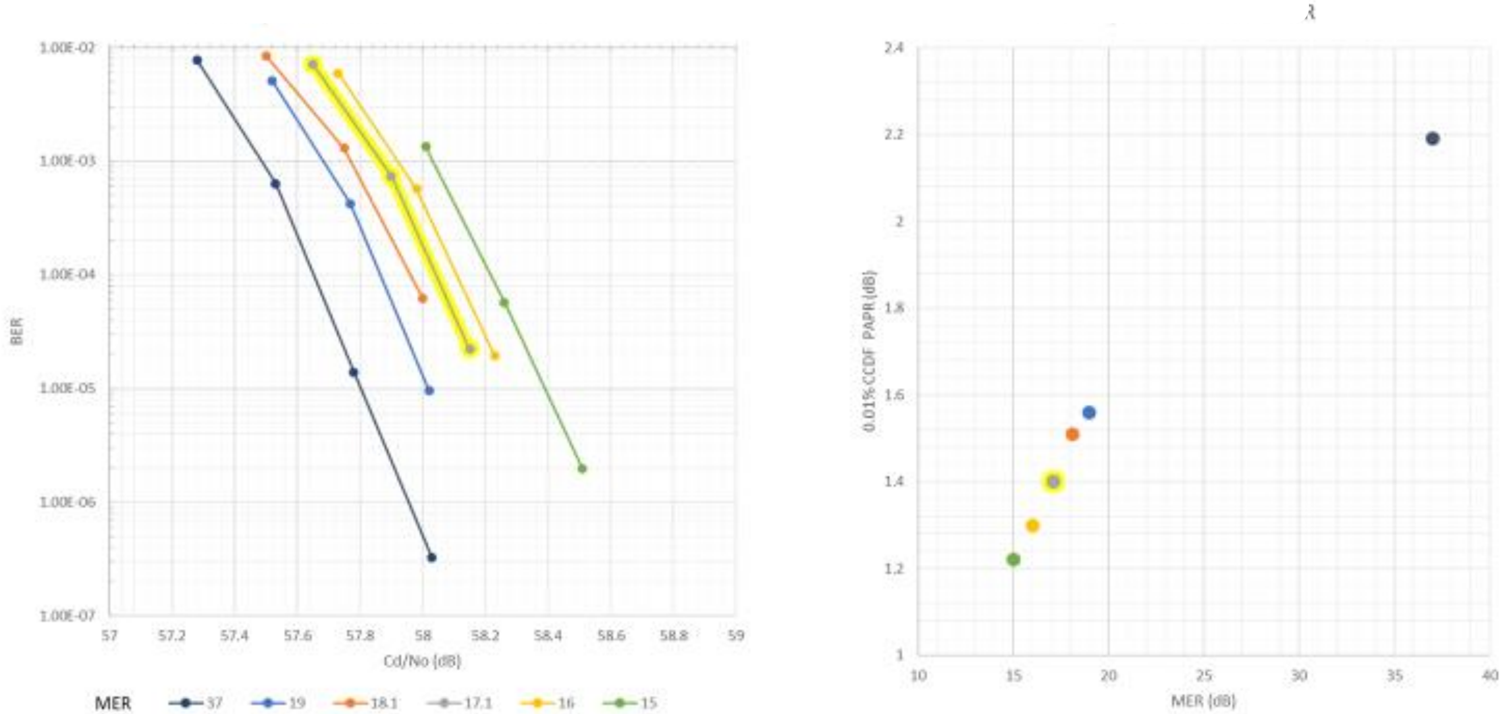


Figure C-4: BER Curves and PAPR Reduction versus MER for Hybrid 16-QAM DSB1 P1

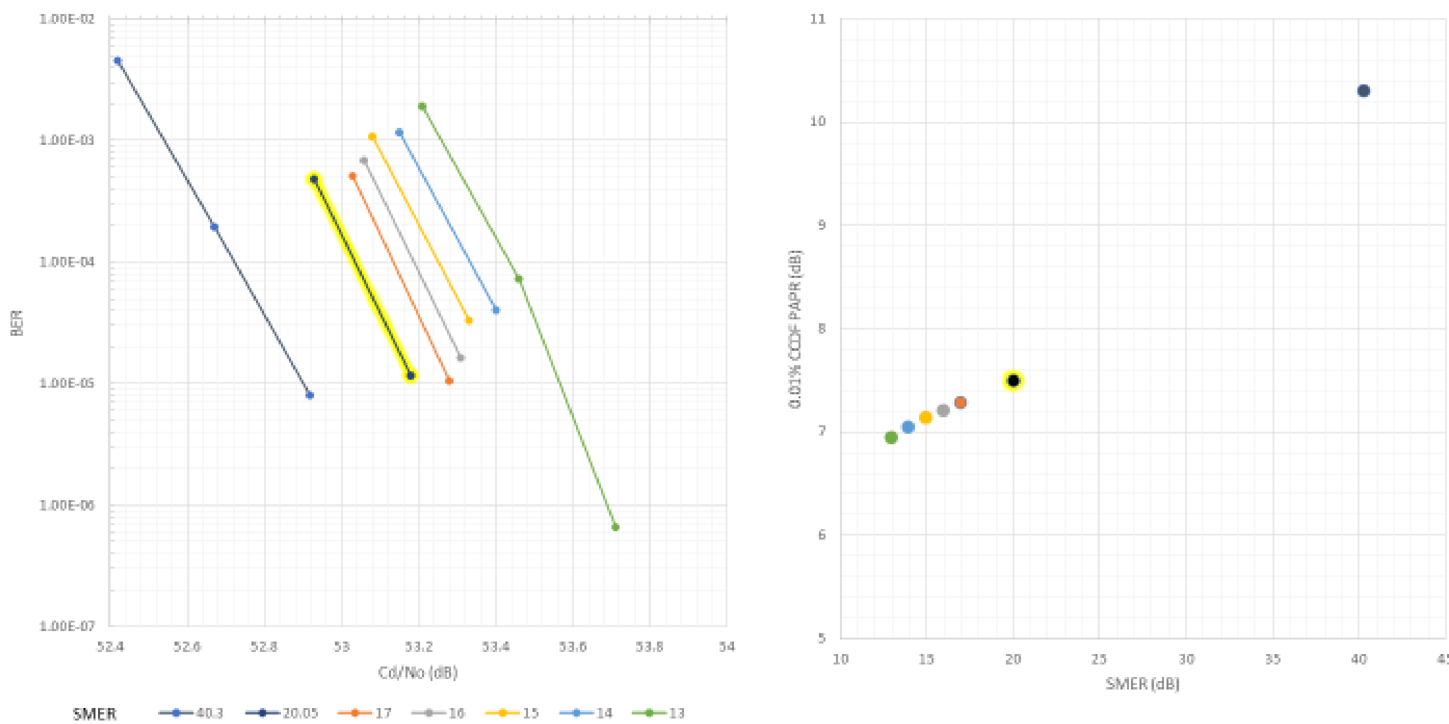
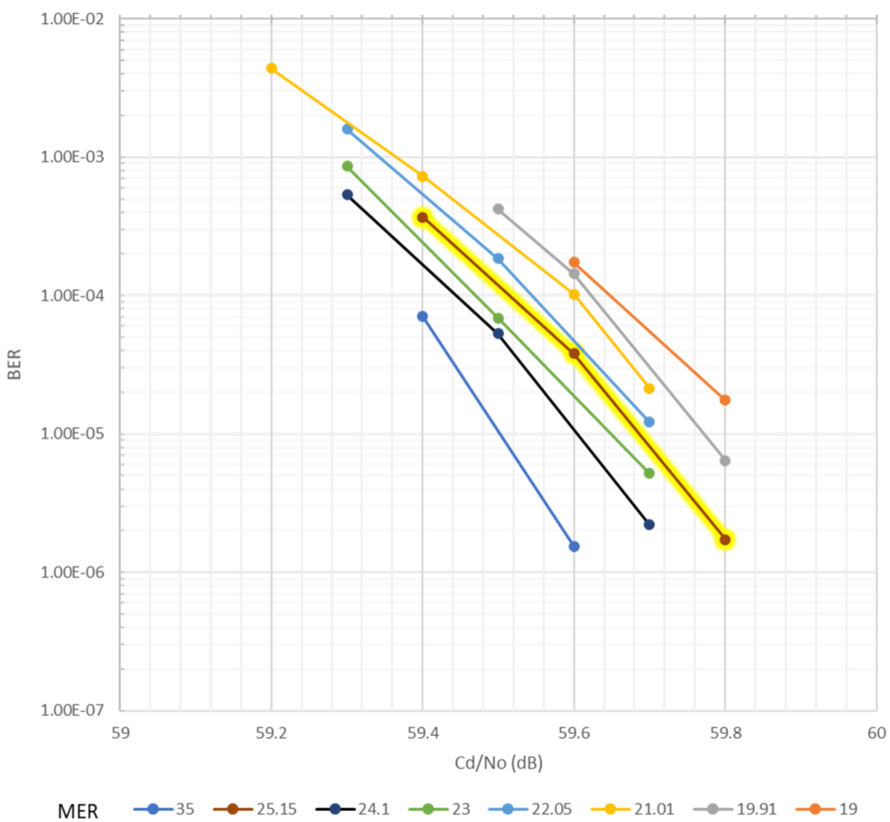


Figure C-5: BER Curves and PAPR Reduction versus MER for All-Digital MS5 S1 @ 10dB Relative to Primary Digital Sidebands



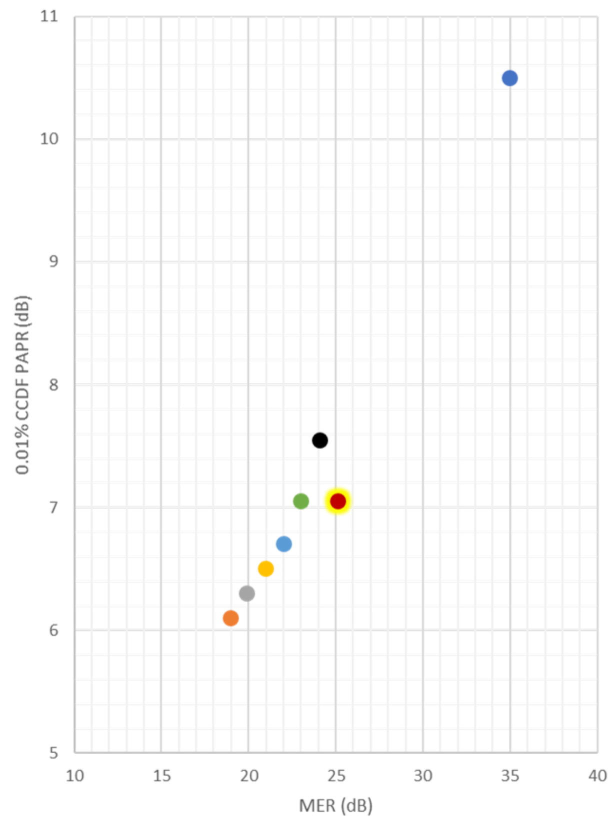


Figure C-6: BER Curves and PAPR Reduction versus MER for Hybrid 64-QAM DSB1OV, P1 Base

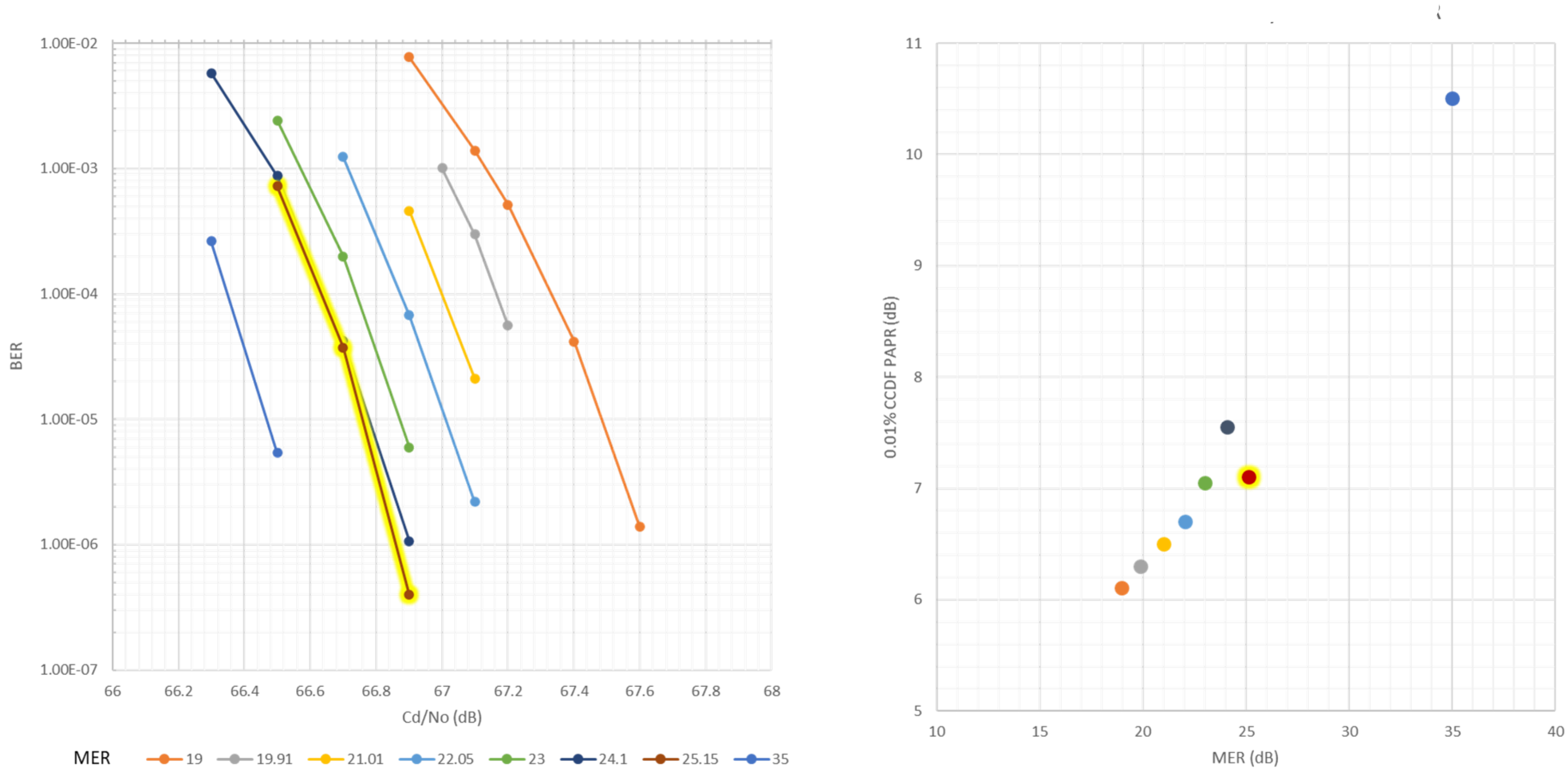


Figure C-7: BER Curves and PAPR Reduction versus MER for Hybrid 64-QAM DSB1OV, POV Overlay

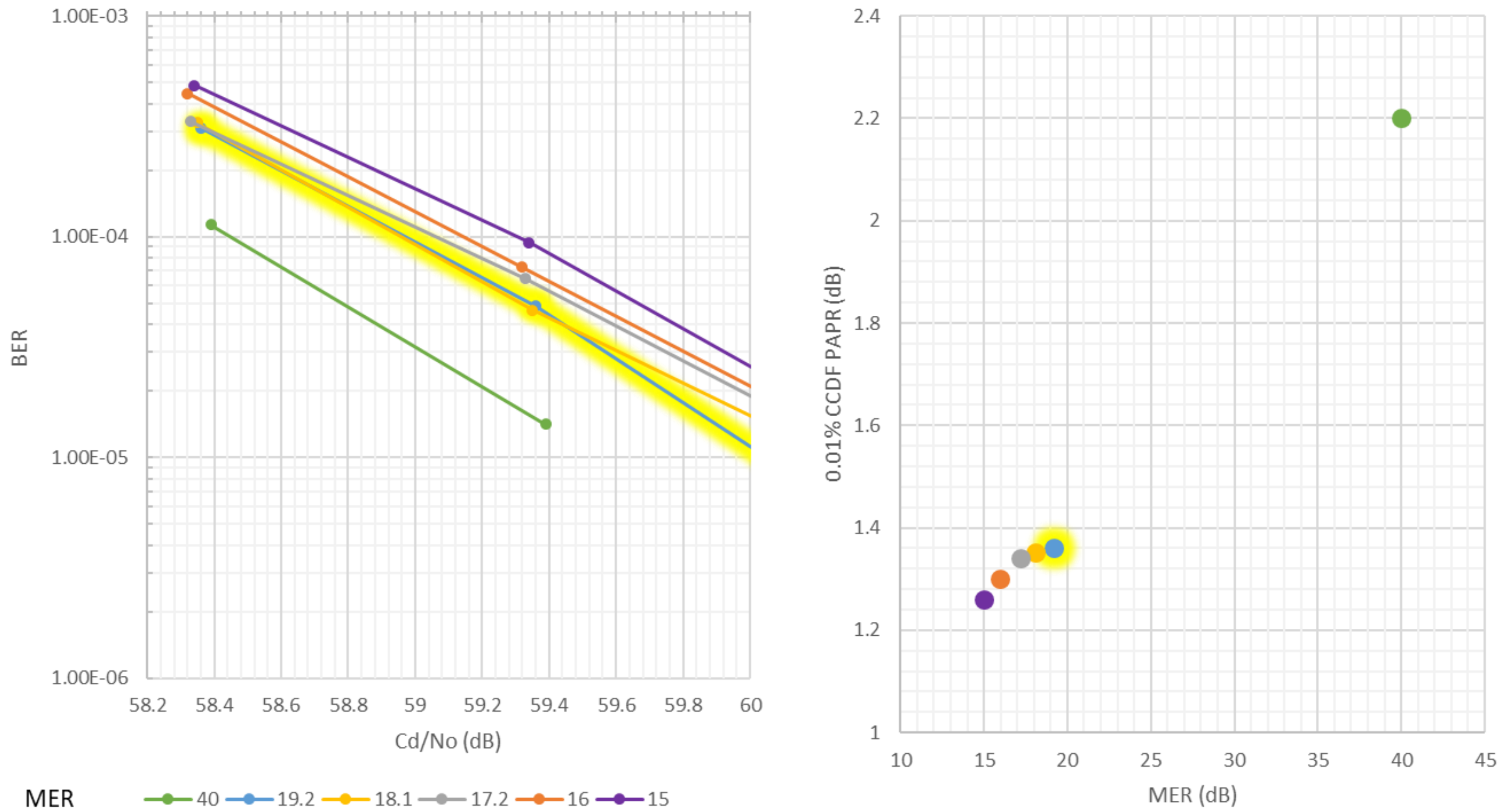


Figure C-8: BER Curves and PAPR Reduction versus MER for Hybrid 16-QAM MP1XOV, P1 Base

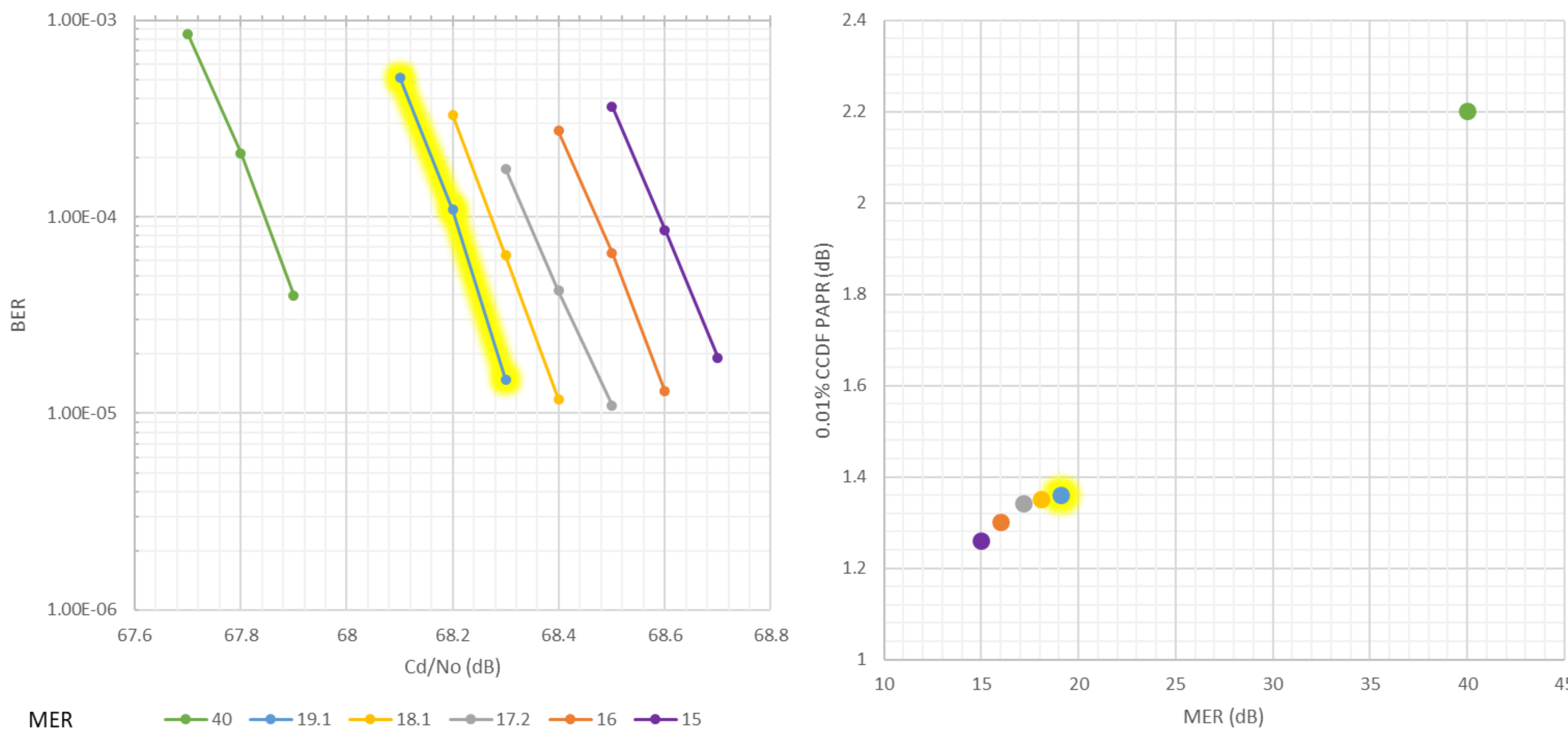


Figure C-9: BER Curves and PAPR Reduction versus MER for Hybrid 16-QAM MP1XOV, POV Overlay

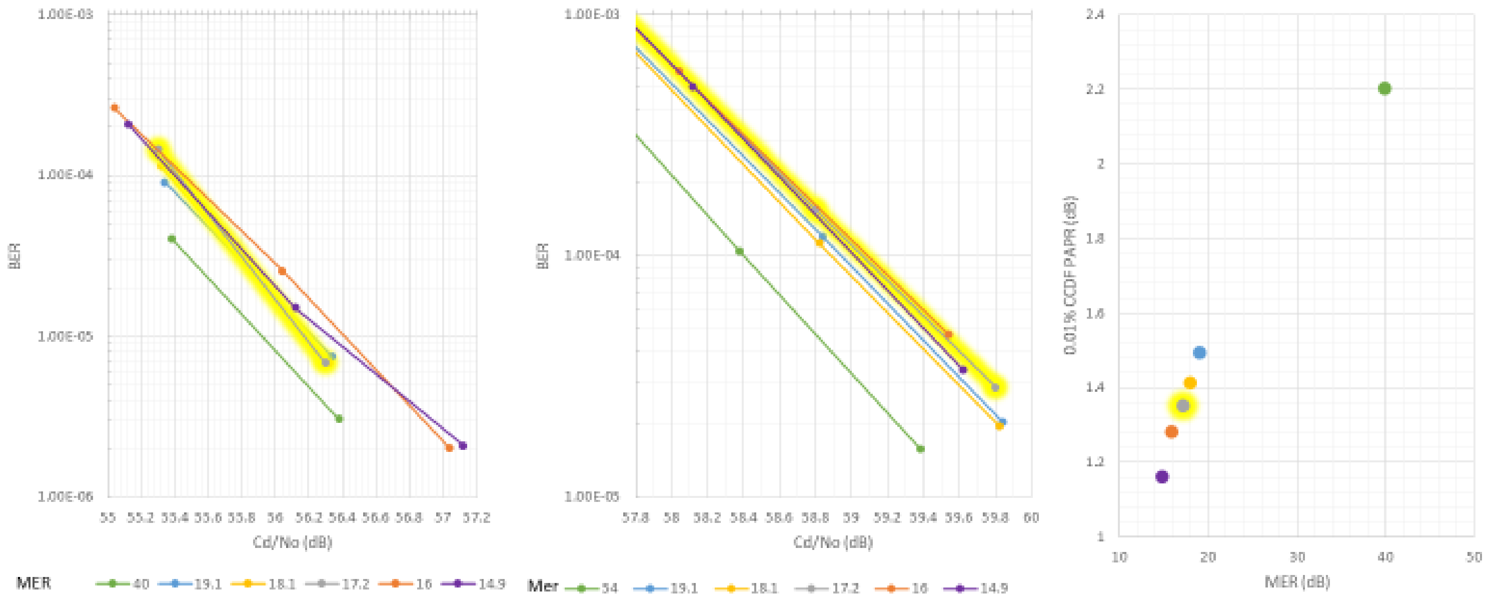
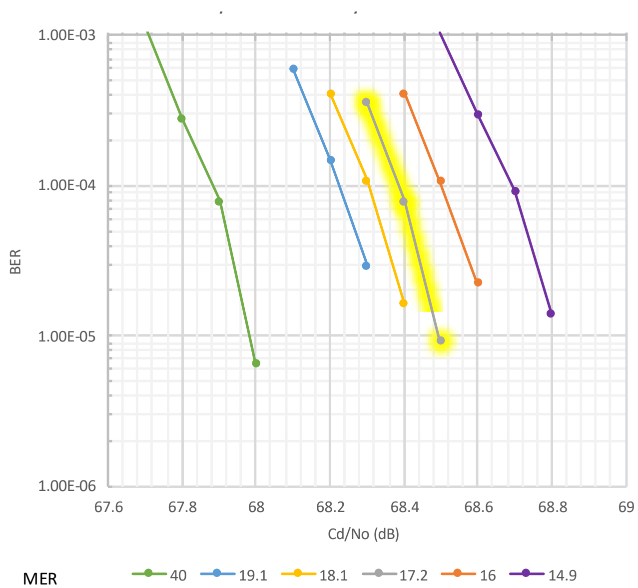


Figure C-10: BER Curves and PAPR Reduction versus MER for Hybrid 16-QAM MP6OV, P1 and P2 Base



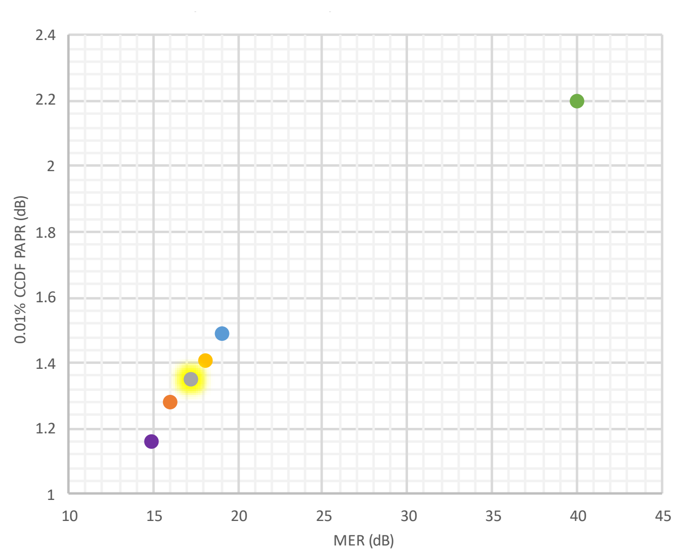


Figure C-11: BER Curves and PAPR Reduction versus MER for Hybrid 16-QAM MP6OV, POV Overlay

Table C-2: Xperi MER Recommendations for Advanced FM Service Modes

Service Mode	MER (dB)	BER Degradation (dB)	PAPR (dB)
MP1 P1	14	0.6	0.9
MP1X P1	14	0.6	1.3
MP1X 16-QAM P4	19	<0.1	1.3
DSB1 Hybrid 16-QAM P1	17	0.4	1.4
DSB1 MS5	20	0.3	7.5
DSB1OV Base	25	0.2	7.1
DSB1OV POV Overlay	25	0.3	7.1
MP1XOV P1/P4 Base	19	0.5	1.4
MP1XOV POV Overlay	19	0.4	1.4
MP6OV Hybrid P1/P2 Base	17	0.6	1.4
MP6OV Hybrid POV Overlay	17	0.5	1.4

Appendix D Peak Power Levels for HD Radio Signal Transmission

D.1 Overview

This appendix presents power measurements for standard and advanced service modes in the FM HD Radio system. These power measurements can help broadcast equipment manufacturers rate their devices for HD Radio broadcasts by estimating the expected peak voltage levels at the input to the device.

Peak voltages can be obtained from the power measurements by scaling the average analog FM voltage of a given transmitter (dependent upon service mode) by the peak power values of Table D-1.

For example, assume that an FM transmitter has an analog transmitter power output of 1000 watts. Assume further that a component at the transmitter output has an input resistance of 50 ohms. So the maximum voltage present at the component input is V = sqrt(50* 1000) = 223.6 V = 47.0 dBV. Assume also that an HD Radio signal in service mode MP1 is being broadcast with PAPR reduction active.

From Table D-1, the peak signal power is expected to be 1.34-dB higher than the analog host signal. Therefore, the peak voltage expected at the input to the component would now be 47.0 dBV + 1.34 dB = 48.3 dBV = 260.9 V.

The measurements presented in this document are subject to the following qualifications:

Results were obtained by simulation. Due to simulation duration limitations, analysis was limited to a real-time equivalent of approximately one (1) hour. Service mode includes future advanced FM service modes. Power levels are expressed in dBm. This provides a reference point for determining the appropriate gains and scaling needed for different power levels. Peak power is not an absolute maximum power, but the maximum power observed within one (1) hour of real-time simulation.

The simulation used Xperi PAPR Reduction algorithm Version 3. The PAPR reduction settings are provided in Table D-2. Settings for the PAPR Reduction algorithm were selected to obtain a specific average transmission sideband MER. The PAPR reduction parameters IT, PD, and CBW listed in Table D-2 are described in Subsection C.1.

D.2 Results from Simulation

Table D-1: Peak Power Measurements from Simulation

Service Mode	Modulation Order		Signal Power (dBm)							0.01% CCDF PAPR (dB)*	
			Analog	Digital			Total Analog + Digital	Peak			
	Primary	Secondary		Primary	Secondary	Total		w/o PAPR Reduction	w/ PAPR Reduction	w/o PAPR Reduction	w/ PAPR Reduction
MP1	QPSK	---	0	-20	---	-20	0.04	3.15	1.34	---	0.91
MP2	QPSK	---	0	-19.6	---	-19.6	0.05	3.25	1.41	---	0.92
MP3	QPSK	---	0	-19.2	---	-19.2	0.05	3.32	1.47	---	0.94
MP1X	QPSK/16QAM	---	0	-18.5	---	-18.5	0.06	3.3	1.55	---	1.06
MP6/MP11	QPSK	---	0	-18.5	---	-18.5	0.06	3.17	1.42	2.04	0.97
DSB1/MP1XOV/MP6 OV	16QAM	---	0	-18.5	---	-18.5	0.06	3.11	1.98	2.04	1.24
DSB1OV	64QAM	---	0	-18.5	---	-18.5	0.06	3.1	2.32	2.03	1.5
MP1X	QPSK/16QAM	---	0	-8.5	---	-8.5	0.57	8.04	4.69	5.01	3.03
DSB1/MP1XOV/MP6 OV	16QAM	---	0	-8.5	---	-8.5	0.57	7.78	5.59	5.01	3.56
DSB1OV	64QAM	---	0	-8.5	---	-8.5	0.57	8.58	7.7	5.01	4.1
DSB1/MP6OV	16QAM	---	---	0	---	0	0	16.76	7.84	9.82	5.03
DSB1OV	64QAM	---	---	0	---	0	0	14.28	10.91	9.81	6.86
DSB1/MS5	16QAM	QPSK	---	-0.41	-10.46	0	0	13.76	9.08	9.82	6.34
DSB1OV/MS5	64QAM	QPSK	---	-0.41	-10.46	0	0	13.44	10.92	9.81	7.77

*These columns indicate that 0.01% of the time the peak-to-average power ratio (PAPR) will exceed the tabulated value

Table D-2: PAPR Reduction Settings Used for Peak Power Measurements

Service Mode	**Modulation Type**		**Signal Power (dBm)**			**PAPR Reduction Setting**												
						IT	**PD**		**CBW**		**Average Sideband MER (dB)**							
											Primary				**Secondary**			
			Analog	**Digital**							**Upper**		**Lower**		**Upper**		**Lower**	
	Prim	**Sec**		**Prim**	**Sec**		**QPSK**	**QAM**	**upper**	**lower**	**QPSK**	**QAM**	**QPSK**	**QAM**	**QPSK**	**QAM**	**QPSK**	**QAM**
MP1	QPSK	---	0	-20	---	4	0.75	---	---	---	14.39	---	14.39	---	---	---	---	---
MP2	QPSK	---	0	-20	---	4	0.75	---	---	---	14.29	---	14.29	---	---	---	---	---
MP3	QPSK	---	0	-20	---	4	0.75	---	---	---	14.21	---	14.20	---	---	---	---	---
MP1X	QPSK 16QAM	---	0	-20	---	4	0.75	0.75	---	---	14.00	18.78	14.00	18.69	---	---	---	---
MP6/MP11	QPSK	---	0	-20	---	4	0.75	---	---	---	14.03	---	14.03	---	---	---	---	---
DSB1 MP1XOV MP6OV	16QAM	---	0	-20	---	8	---	0.75	---	---	---	17.69	---	17.69	---	---	---	---
DSB1OV	64QAM	---	0	-20	---	8	---	0.75	---	---	---	24.11	---	24.12	---	---	---	---
MP1X	QPSK 16QAM	---	0	-10	---	4	0.75	0.75	---	---	14.20	18.15	14.20	18.16	---	---	---	---
DSB1 MP1XOV MP6OV	16QAM	---	0	-10	---	8	---	0.77	---	---	---	18.29	---	18.29	---	---	---	---
DSB1OV	64QAM	---	0	-10	---	8	---	0.73	---	---	---	23.38	---	23.37	---	---	---	---
DSB1 MP6OV	16QAM	---	---	0	---	8	---	0.69	---	---	---	18.31	---	18.31	---	---	---	---
DSB1OV	64QAM	---	---	0	---	8	---	0.72	---	---	---	23.20	---	23.20	---	---	---	---
DSB1 MS5	16QAM	QPSK	---	-0.41	-10.46	8	0.75	0.75	0.38	0.38	---	17.95	---	17.96	14.14	---	14.14	---
DSB1OV MS5	64QAM	QPSK	---	-0.41	-10.46	8	0.75	0.73	0.43	0.43	---	23.50	---	23.50	13.95	---	13.95	---

D.3 Summary

This appendix provides simulated peak power values for various FM HD Radio service modes – including advanced FM service modes – with and without PAPR reduction. These peak power values can be used by broadcast equipment manufacturers to determine the peak voltage ratings for their components.

Table D-2 shows that all-digital advanced FM service modes with higher-order modulation produce the highest expected signal peaks. Because the analog host signal is no longer present, the superposition of the OFDM digital subcarriers dominates the output signal characteristics. If PAPR reduction is employed by the transmitter, the peak power levels are effectively reduced.

With the transmission of advanced FM service modes, broadcast equipment manufacturers must consider the increasing variety of potential HD Radio signal types – and their associated peak power levels – when developing their products.

Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard

Revision 03
November 2022

SY_REF_2690s

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1 Scope

1.1 System Overview

iBiquity Digital Corporation's 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

iBiquity Digital Corporation's reference documents that support the description of the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard are listed in this document.

2 Reference Documents

	Company / Document Title	Document / Revision
[1]	iBiquity Digital Corporation “HD Radio™ Air Interface Design Description – Layer 1 FM”	SY_IDD_1011s Revision H
[2]	iBiquity Digital Corporation “HD Radio™ Air Interface Design Description – Layer 1 AM”	SY_IDD_1012s Revision G
[3]	iBiquity Digital Corporation “HD Radio™ Air Interface Design Description – Layer 2 Channel Multiplex”	SY_IDD_1014s Revision K
[4]	iBiquity Digital Corporation “HD Radio™ Air Interface Design Description – Audio Transport”	SY_IDD_1017s Revision I
[5]	iBiquity Digital Corporation “HD Radio™ Air Interface Design Description – Advanced Application Services Transport”	SY_IDD_1019s Revision H
[6]	iBiquity Digital Corporation “HD Radio™ Air Interface Design Description – Station Information Service”	SY_IDD_1020s Revision K
[7]	iBiquity Digital Corporation “HD Radio™ FM Transmission System Specifications”	SY_SSS_1026s Revision H
[8]	iBiquity Digital Corporation “HD Radio™ Air Interface Design Description – Program Service Data”	SY_IDD_1028s Revision E
[9]	iBiquity Digital Corporation “HD Radio™ AM Transmission System Specifications”	SY_SSS_1082s Revision H
[10]	iBiquity Digital Corporation “HD Radio™ Air Interface Design Description – Program Service Data Transport”	SY_IDD_1085s Revision D
[11]	Federal Communications Commission (FCC) “Code of Federal Regulations”, Title 47, Part 11	~
[12]	Federal Communications Commission (FCC) “Code of Federal Regulations”, Title 47, Part 73	~
[13]	National Radio Systems Committee (NRSC) “NRSC AM Pre-emphasis/De-emphasis and Broadcast Audio Transmission Bandwidth Specifications” NRSC-1-B, September 2012	~
[14]	International Organization for Standardization (ISO) “English Country Names and Code Elements” ISO 3166-1 and corresponding ISO 3166-1-alpha-2 Code Elements Document published on: 2013-11-15	~

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[15]	United States Department of Transportation "Uniform Time Act of 1966" Public Law 89-387, April 13, 1966, 80 Statute 107, 15 U.S.C §260a	~
[16]	United States Congress (United States Department of Energy) "Energy Policy Act of 2005" Public Law 109-058, August 8, 2005.	~
[17]	United States Federal Communications Commission (FCC) "Media Bureau Consolidated Database System (MB CDBS)" Web URL: http://www.fcc.gov/mb/cdbs.html 2016 Web Links https://www.fcc.gov/media/filing-systems-and-databases http://licensing.fcc.gov/prod/cdbs/pubacc/prod/cdbs_pa.htm	~
[18]	United States National Geospatial-Intelligence Agency (NGA) "Department of Defense World Geodetic System 1984, Its Definition and Relationships with Local Geodetic Systems" Third Edition, 4 July 1997, NIMA Technical Report TR8350.2 Web URL: http://earth-info.nga.mil/GandG/publications/tr8350.2/tr8350_2.html 2016 Web Links http://earth-info.nga.mil/GandG/publications/tr8350.2/tr8350_2.html	~
[19]	United States National Institute of Standards and Technology (NIST) "Information about the new Daylight Saving Time (DST)" Web URL: http://tf.nist.gov/timefreq/general/dst.htm 2016 Web Links http://www.nist.gov/pml/div688/dst.cfm https://www2.nist.gov/time-and-frequency-division/time-and-frequency-division-popular-links/daylight-saving-time-dst	~

	Company / Document Title	Document / Revision
[20]	EU European Parliament and Council “Proposal for a European Parliament and Council directive on summer-time arrangements” Web URL: http://europa.eu/bulletin/en/200012/p104047.htm 2016 Web Links http://ec.europa.eu/transport/summertime_en.htm http://ec.europa.eu/transparency/regdoc/rep/1/2007/EN/1-2007-739-EN-F1-1.Pdf Directive 2000/84/EC of the European Parliament and of the Council of 19 January 2001 on summer-time arrangements	~
[21]	International Earth Rotation and Reference Systems Service (IERS) “Bulletin C - Announcement of Leap Seconds in UTC” Web URL: http://www.iers.org/ 2016 Web Links https://www.iers.org/IERS/EN/Home/home_node.html	~
[22]	International Organization for Standardization (ISO) “Information Technology - 8-bit single-byte coded graphic character sets - Part 1: Latin Alphabet 1” ISO/IEC 8859-1:1998. Web URL: http://www.iso.org/iso/iso_catalogue/catalogue_tc/catalogue_detail.htm?csnumber=28245 2016 Web Links http://www.iso.org/iso/iso_catalogue/catalogue_tc/catalogue_detail.htm?csnumber=28245	~
[23]	International Organization for Standardization (ISO) “Information Technology - Universal Multiple-Octet Coded Character Set (UCS) - Part 1: Architecture and Basic Multilingual Plane” ISO/IEC 10646-1:2000. Web URL: http://www.iso.org/iso/iso_catalogue/catalogue_tc/catalogue_detail.htm?csnumber=29819 2016 Web Links http://www.iso.org/iso/iso_catalogue/catalogue_tc/catalogue_detail.htm?csnumber=29819	~
[24]	RFC 1662, Network Working Group “PPP in HDLC-like Framing” Web URL: http://www.ietf.org/rfc/rfc1662.txt 2016 Web Links | http://www.ietf.org/rfc/rfc1662.txt	~

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[25]	Martin Nilsson “ID3v2.3.0 Informal standard” Web URL: http://www.id3.org 2016 Web Links | http://id3.org/id3v2.3.0	~
[26]	National Radio Systems Committee (NRSC) NRSC-G201-A, “NRSC-5 RF Mask Compliance: Measurement Methods and Practice”, April 2010.	~
[27]	iBiquity Digital Corporation “Transmission Signal Quality Metrics for FM IBOC Signals”	SY_TN_2646s Revision 03
[28]	Reserved	Reserved
[29]	National Radio Systems Committee (NRSC) NRSC-4-B, “United States RBDS Standard: Specification of the radio broadcast data system (RBDS)”, April 2011.	~
[30]	National Radio Systems Committee (NRSC) “Bandwidth Options for Analog AM Broadcasters” NRSC-G100-A, September 2012.	~
[31]	OASIS “Common Alerting Protocol (CAP) Version 1.2”. Web URL: http://docs.oasis-open.org/emergency/cap/v1.2/CAP-v1.2.pdf 2016 Web Links | http://docs.oasis-open.org/emergency/cap/v1.2/CAP-v1.2.pdf	~

	Company / Document Title	Document / Revision
[33]	International Organization for Standardization (ISO) ISO/TS 18234-3:2013 Intelligent transport systems -- Traffic and travel information via transport protocol experts group, generation 1 (TPEG1) binary data format -- Part 3: Service and network information (TPEG1-SNI) Document published on: 2013-02-01	~
[34]	DTS, Inc. “NRSC Supplemental Information” Web URL: http://hdradio.com/broadcasters/us-regulatory/nrsc-supplemental-information	~
[35]	National Radio Systems Committee (NRSC) “Harmonization of RDS and IBOC Program Service Data (PSD) Guideline” NRSC-G200-A, March 2010.	~
[36]	Reserved	Reserved
[37]	iBiquity Digital Corporation “HD Radio Air Interface Design Description Low-Latency Data Service Transport”	SY_IDD_4363s Version A
[38]	National Radio Systems Committee (NRSC) “IBOC Time and Level Alignment Guideline” NRSC-G203-A, September 2021.	~

HD Radio™ Air Interface Design Description Low-Latency Data Service Transport

Version A

November 2022

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1.2 Document Overview

This document describes how control information and data are passed through the Low-Latency Data Service (LLDS) Transport for subsequent processing by Layer 2.

2 Referenced Documents

STATEMENT
Each referenced document that is mentioned in this document shall be listed in the following iBiquity document:

- Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard Document Number: SY_REF_2690s

3 Abbreviations and Conventions

3.1 Abbreviations and Acronyms

ADV	Advanced Processing
ALFN	Absolute L1 Frame Number
AM	Amplitude Modulation
ASCII	American Standard Code for Information Interchange
CAP	Common Alerting Protocol
CRC	Cyclic Redundancy Check
DST	Daylight Saving Time
EA	Emergency Alerts
EBU	European Broadcasting Union
FCC	Federal Communications Commission
FEMA	Federal Emergency Management Agency
FM	Frequency Modulation
GPS	Global Positioning System
IBOC	In-Band On-Channel
ID	Identification
ID3	Tag Embedded in MPEG I Layer III Files
ISO	International Organization for Standardization
L1	Layer 1
LSB	Least Significant Bit
LLDS	Low-Latency Data Service
MF	Medium Frequency
MIME	Multipurpose Internet Mail Extensions
MSB	Most Significant Bit
MSG	Message
PDU	Protocol Data Unit
PIDS	Primary IBOC Data Service Logical Channel
PIDSOV	Primary IBOC Data Service Overlay Logical Channel
RDS	Radio Data System
SIDS	Secondary IBOC Data Service Logical Channel
SIS	Station Information Service
UTC	Coordinated Universal Time
VHF	Very High Frequency
WGS	World Geodetic System

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 most 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 lowest index.
- In representations of binary numbers, the least significant bit is on the right.
- Hexadecimal numbers are represented by a prefix of “0x”

4 Low-Latency Data Service Protocol Data Unit Format

The Low-Latency Data Service (LLDS) provides data and control information for applications that require timely delivery of relatively modest amounts of data. LLDS is supported by most advanced FM service modes and by none of the standard service modes.

The information may be transmitted in a series of LLDS Protocol Data Units (PDUs) on the Primary IBOC Data Service (PIDS) logical channel in service modes DSB1, and DSB1OV. These advanced FM service modes provide expanded PIDS capacity beyond the 10 bytes per L1 block afforded to the Station Information Service (SIS) transport [6] in standard FM service modes. Any PIDS capacity over and above these 10 SIS bytes per L1 block is available for additional SIS transport PDUs and/or LLDS transport PDUs. For more information on PIDS, see [1] and [2].

Furthermore, the PIDSOV logical channel in advanced FM layered service modes (MP1XOV, MP6OV, and DSB1OV), and the SIDS logical channel in all-digital waveforms (secondary service mode MS5), may also be used by the SIS and LLDS transports.

The high-level LLDS PDU format is shown in Figure 4-1. Since LLDS shares the PIDS, PIDSOV, and SIDS logical channels with the SIS transport, the high-level 80-bit PDU format is common to that of the SIS. The Type bit is set to one to indicate LLDS. SIS PDUs have the type bit set to zero. The CRC field is the same for both. This commonality simplifies multiplexing PIDS, PIDSOV, and SIDS capacity between the two services.

The most significant bit of each field is shown on the left. Layer 2 and Layer 1 process MSBs first; that is, bit 0 is the first bit interleaved by L1. The PDU contents are defined by several control fields within the PDU.

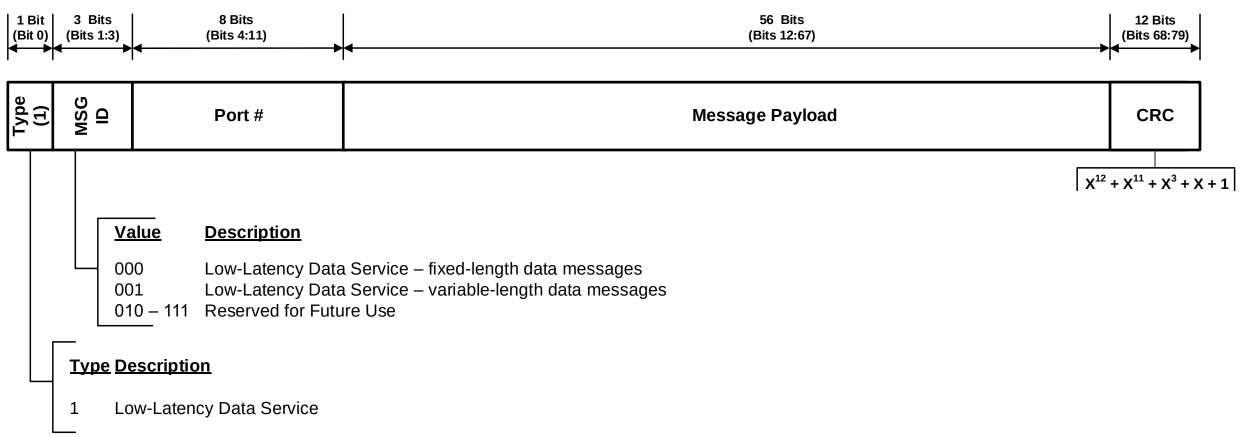

Figure 4-1: LLDS PDU Format

Up to 254 different LLDS data ports may be defined. Each one is specified by an 8-bit port number. Port assignments may be arbitrary: i.e., data services are uniquely identified to a receiver by their MIME hash value elsewhere, not by their port number. Refer to [6]. For a given broadcast, each data service must be assigned to a unique port number. But port numbers are free to be re-used by other broadcasters.

LLDS ports share the same 16-bit port numbering scheme with the AAS transport [5]. The most significant byte of LLDS data ports is always 0xFF. Note that due to the limited capacity of LLDS, only a small number of LLDS services may be simultaneously supported by one broadcast.

The message ID defines the specific LLDS message type as defined in Table 4-1. Currently only two types are defined. Others may be defined in the future. Type 0 messages consist of 7 bytes of LLDS data. No other protocol information is provided. It is assumed that additional control information will be defined by the service provider. Fixed-length LLDS data messages are most useful for sending very short data messages that do not require the additional overhead incurred by Type 1 messages.

Type 1 messages are variable length, with a maximum LLDS data length of 190 bytes. Messages typically span multiple PDUs. Additional overhead is provided to support this feature, so this message type is preferred to support longer messages.

Table 4-1: LLDS PDU MSG ID Definitions

MSG ID	Payload Size (bits)	Description	Comments
000	56	LLDS Fixed-Length Data Message	Transports 7 bytes of raw data
001	56	LLDS Variable-Length Data Message	Includes additional overhead to support data messages varying in size up to 190 bytes in length
010 - 111	56	Reserved	Reserved for Future use

The following subsections describe each LLDS message type (MSG ID).

4.1 LLDS Fixed-Length Data Message (MSG ID = 000)

This message type is used to send 7 bytes of LLDS data. The service provider is free to uniquely define the high-level protocol and packet structure for their intended service. Figure 4-2 shows the message structure.

Figure 4-2: LLDS Fixed-Length Data Message Structure

Table 4-2 describes the LLDS Fixed-Length Data Message fields.

Table 4-2: LLDS Fixed-Length Data Message – Field Names and Descriptions

Field Name	Number of Bits	Field Description
Port Number	8	Defines the data port number for the service. LLDS data ports share a common 16-bit port numbering scheme with AAS data ports described in reference [5]. LLDS Data ports have the most significant byte of the port number set to 0xFF. i.e., LLDS port numbers range from 0xFF00 to 0xFFFF, where the least significant byte is reported in this field. Port numbers 0xFF7D and 0xFF7E are invalid and not to be used.
LLDS Data	56	7 bytes of LLDS data The packet format is defined by the service provider.

4.2 LLDS Variable-Length Data Message (MSG ID 001)

This message type is used to send variable-length data messages up to 190 bytes in length.

LLDS Variable-Length Data Messages typically span multiple frames/LLDS PDUs. Each LLDS PDU has a total payload of 56 bits.

The structure for LLDS Variable-Length Data Messages is shown in Figure 4-3. The format of frame n = 0 is different from frames n = 1 to n = 31, as shown.

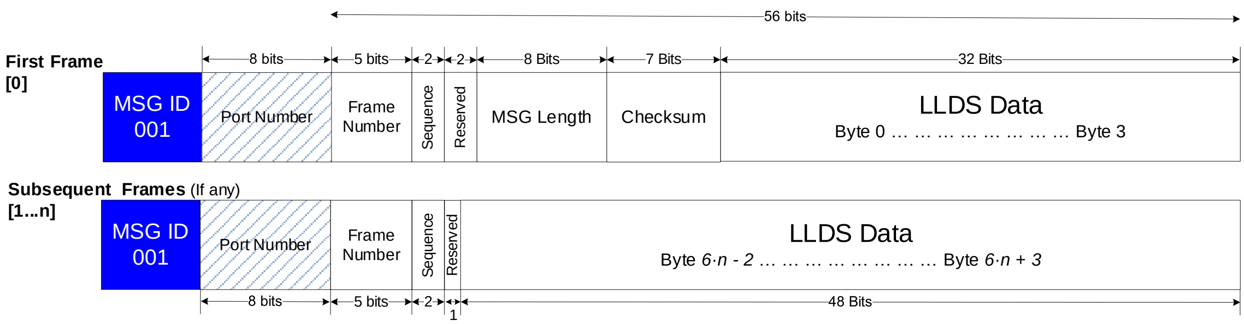

Figure 4-3: LLDS Variable-Length Data Message Structure

An LLDS Variable-Length Data Message may span up to 32 frames. Each data message frame contains a sequence number indicating when the overall data message has changed.

Any change in the data message content is considered a new data message, and the sequence number is incremented. A 7-bit checksum is included in frame 0 to improve the reliability of reception.

Table 4-3 and Table 4-4 describe the LLDS Variable-Length Data Message fields for frame 0 and subsequent frames, respectively.

Table 4-3: LLDS Variable-Length Data Message – Field Names and Descriptions for Frame Number n = 0

Field Name	Range	Description
Port Number	0-255, excluding 125 and 126	Defines the data port number for the service. LLDS data ports share a common 16-bit port numbering scheme with AAS data ports described in reference [5]. LLDS Data ports have the most significant byte of the port number set to 0xFF. i.e., LLDS port numbers range from 0xFF00 to 0xFFFF where the least significant byte is reported in this field. Port numbers 0xFF7D and 0xFF7E are invalid and not to be used.
Frame Number (n)	0	Indicates the current frame number of the LLDS Variable-Length Data message. Set to zero for the first frame.

Field Name	Range	Description
Sequence	0 - 3	Increments by 1, modulo 4, whenever the data message changes. A new sequence number must commence with frame 0 and the same number shall be used for all frames of a given data message. Multiple repeats of the same data message are allowed.
Reserved	0-3	These two bits are reserved for future use. Shall be set to zero by default.
MSG Length	4 - 190	Defines the total number of bytes in the LLDS Data fields across all frames.
Checksum	0 - 127	Checksum of all the bytes in the LLDS Data fields across all frames. Refer to Figure 4-4 for details.
LLDS Data	N/A	Frame 0 contains the first 4 bytes of the data message. The packet format of the LLDS data is defined by the service provider. Byte 0 is the leftmost byte. For single-frame data messages, any unused bytes to the right of the actual data are filled with NULL characters (0x00).

Table 4-4: LLDS Variable-Length Data Message – Field Names and Descriptions for Frame Numbers n = 1 to 31

Field Name	Range	Description
Port Number	0-255, excluding 125 and 126	Defines the data port number for the service. LLDS data ports share a common 16-bit port numbering scheme with AAS data ports described in reference [5]. LLDS Data ports have the most significant byte of the port number set to 0xFF. i.e., LLDS port numbers range from 0xFF00 to 0xFFFF where the least significant byte is reported in this field. Port numbers 0xFF7D and 0xFF7E are invalid and not to be used.
Frame Number (n)	1 - 31	Indicates the current frame number of the LLDS Variable-Length Data message.
Sequence	0 - 3	Increments by 1, modulo 4, whenever the data message changes. A new sequence number must commence with frame 0 and the same number shall be used for all frames of a given data message. Multiple repeats of the same data message are allowed.
Reserved	0 - 1	This bit is reserved for future use. Shall be set to zero by default.
LLDS Data	N/A	Frames 1 to n contain the remaining bytes of the data message, where the lowest numbered byte within a frame is the leftmost for that frame. For the last frame, any unused bytes to the right of the data message bytes are filled with NULL characters (0x00).

4.2.1 LLDS Variable-Length Data Message Checksum

Figure 4-4 illustrates the method used to calculate the 7-bit checksum contained in frame 0. First, a 16-bit sum is computed by adding together all bytes sent in the LLDS Data fields across all frames of the LLDS Variable-Length Data message.

The LLDS Data bytes and the sum are both treated as unsigned integers. The 16-bit sum is then split into a high (most significant) byte and a low (least significant) byte. The most significant bit of the high byte (bit 15 in Figure 4-4) is cleared. The high and low bytes are then summed, and the seven least significant bits of the sum are written into the checksum field, where the most significant bit is the left-most checksum bit shown in Figure 4-3.

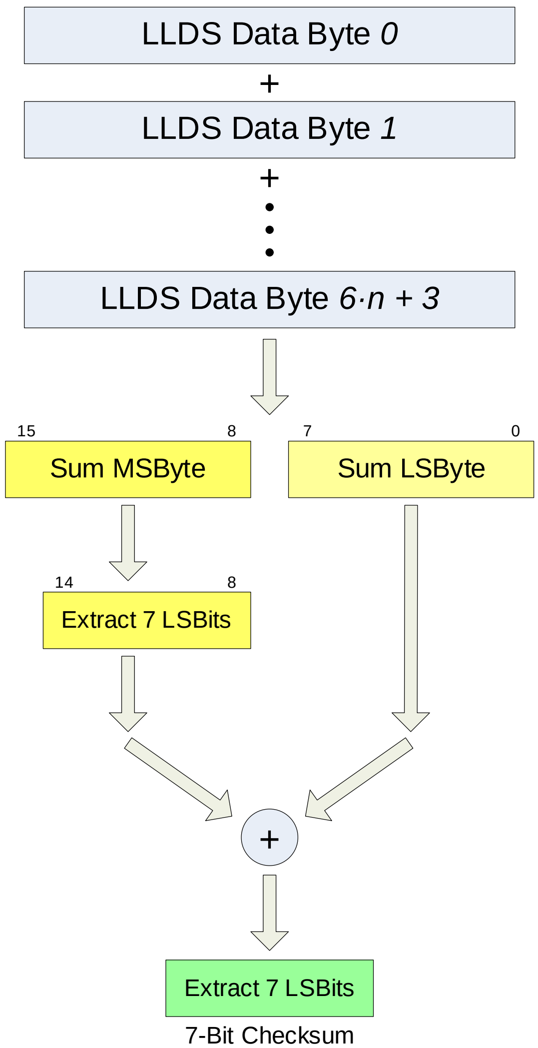

Figure 4-4: LLDS Variable-Length Data Message Checksum Field Calculation

4.3 LLDS PDU CRC Field

Regardless of message type, each PDU is terminated with a 12-bit Cyclic Redundancy Check (CRC) for the purpose of aiding the receiver in detecting transmission errors. The CRC, ordered as PDU bits 79:68, is computed as follows:

1. Fill PDU bits 79:68 with zeros.
2. Perform modulo-two division of PDU bits 79:0 by the generator polynomial g(x),

   Where $g(x) = X^{12} + X^{11} + X^{3} + X + 1$

   and PDU bit 79 is computed first.
3. The 12-bit remainder is then copied back into PDU bits 68:79, where bit 68 is considered the most significant remainder bit and bit 79 is the least significant remainder bit.

5 LLDS PDU/Message Scheduling

The LLDS transport shares PIDS, PIDSOV, and SIDS data capacity with the SIS transport [6].

The two transports are multiplexed onto the Layer 1 PIDS logical channel. In addition, some of the advanced FM service modes also provide PIDSOV and/or SIDS logical channels that afford separate capacity for additional SIS and LLDS messages. Refer to [1] for details.

LLDS message scheduling is flexible as long as a certain minimum PIDS capacity is dedicated to SIS messages. Refer to Table 5-1 for details. Note that only some of the advanced FM service modes support LLDS. SIS requires full PIDS capacity in the standard FM service modes. The primary reason for this is that SIS supports time-critical delivery of both Emergency Alerts as well as Service Information Messages that support receiver audio and background scanning functions.

Table 5-1: LLDS PIDS PDU Scheduling Constraints

Service Mode	Total # of PIDS Channel PDUs per L1 Frame	Minimum # of SIS PDUs Required per L1 Frame	Maximum # of LLDS PDUs Available per L1 Frame
MA1, MA3	8	8	0 LLDS not supported by these modes
MP1, MP1X, MP1XOV, MP2, MP3, MP5, MP6, MP6OV, MP11	16	16	0 LLDS not supported via PIDS for these modes
DSB1, DSB1OV	64	16	48

The PIDS scheduling constraints in Table 5-1 apply to every L1 frame. This is necessary for receivers scanning for audio and/or data services to meet their pre-programmed timing specifications and to optimize the user experience. As long as the per-L1-frame SIS requirements are met, LLDS messages may be sent in any available timeslots. However, service providers are cautioned to minimize LLDS usage as much as possible due to its limited data capacity coupled with the need to keep latency low. LLDS should be considered only for applications where the delays incurred by the AAS transport are prohibitive.

Note that as advanced FM service modes are rolled out and stations offer more audio and data services afforded by the higher overall system capacity, SIS PDU capacity may need to be increased to support receiver scan times for these additional services.

Some advanced FM service modes use layered (hierarchical) modulation, which is a backward-compatible means of enhancing capacity by simultaneously broadcasting two service modes, one of which is “overlayed” on top of an existing “base” service mode. Legacy receivers detect the base layer, while newer advanced receivers also detect the overlay for additional capacity. In addition to the PIDS logical channel carried by the base service mode, layered service modes also provide a PIDSOV logical channel on the overlay. The intended application of the PIDSOV logical channel is the same as the PIDS logical channel, providing additional capacity for SIS and LLDS services, as shown in Table 5-2.

Table 5-2: LLDS PIDSOV PDU Scheduling Constraints

Service Mode	**Total # of PIDSOV Channel PDUs per L1 Frame**	**Minimum # of SIS PDUs Required per L1 Frame**	**Maximum # of LLDS PDUs Available per L1 Frame**
MP1XOV, MP6OV, DSB1OV	32	0	32

All-digital FM waveforms broadcasting secondary service mode MS5 may also use the SIDS logical channel to provide even more capacity for SIS and LLDS services, as shown in Table 5-3.

Table 5-3: LLDS SIDS PDU Scheduling Constraints

Service Mode	**Total # of SIDS Channel PDUs per L1 Frame**	**Minimum # of SIS PDUs Required per L1 Frame**	**Maximum # of LLDS PDUs Available per L1 Frame**
MS5	32	0	32

A simple PIDSOV scheduling example is shown in Figure 5-1. The PIDSOV logical channel has 16 L1 blocks per L1 frame, numbered from 0 to 15. In the example (for layered service modes MP1XOV, MP6OV, or DSB1OV), there are two 80-bit PDUs per L1 block, for a total of 32 PDUs per L1 frame.

As shown in Figure 5-1, the first PDU of every block is dedicated to SIS messages. In the second PDU of every block, there are three multiplexed LLDS services. LLDS Fixed-Length Data Messages are sent on port 0xFF01. On Port 0xFF14, frame numbers 1, 2, and 3 of an LLDS Variable-Length Data Message are sent between blocks 1 and 5. In block 6, frame 0 of a new message is sent, and the sequence number is incremented from 1 to 2. The sequence 2 message has a length of two frames. In block 14, a new message is sent, and the sequence number is incremented again. On port 0xFF87, the last 3 frames of an LLDS Variable-Length Data Message are shown, spanning blocks 0, 4, and 8. In block 10, a new message is sent, and the sequence number is incremented (modulo 4) to 0.

L1 Block #	PDU #1	PDU #2	
0	SIS	3:3	Port 0xFF87
1	SIS	1:1	Port 0xFF14
2	SIS	2:1	Port 0xFF14
3	SIS		Port 0xFF01
4	SIS	4:3	Port 0xFF87
5	SIS	3:1	Port 0xFF14
6	SIS	0:2	Port 0xFF14
7	SIS		Port 0xFF01
8	SIS	5:3	Port 0xFF87
9	SIS	1:2	Port 0xFF14
10	SIS	0:0	Port 0xFF87
11	SIS		Port 0xFF01
12	SIS	1:0	Port 0xFF87
13	SIS	2:0	Port 0xFF87
14	SIS	0:3	Port 0xFF14
15	SIS		Port 0xFF01

Key	
(black)	SIS Message
(dark gray)	LLDS Port 0xFF01 Fixed-Length Data Message
(light gray)	LLDS Port 0xFF14 Variable-Length Data Message
(very light gray)	LLDS Port 0xFF87 Variable-Length Data Message

Variable-Length Data Message data is labelled X:Y, where X = MSG Frame #, Y = Sequence #

Figure 5-1: SIS/LLDS Scheduling Example – 2 PIDSOV PDUs/Block

Figure 5-2 shows an example of 4 PIDS PDUs per block, as would be the case for advanced service modes DSB1 and DSB1OV. In this case, 24 PIDS PDUs per frame are allocated to SIS, providing extra capacity to support the additional audio and data services provided by a higher-capacity service mode. LLDS Variable-Length Data Messages are sent on port 0xFF11 in PDU #4 of every block.

A second LLDS Variable-Length Data Message service is sent on port 0xFF62 in some of the blocks of both PDUs #2 and #3. The remaining capacity is used by an LLDS Fixed-Length Data Message service on port 0xFF29.

L1 Block #	PDU #1	PDU #2	PDU #3	PDU #4
0	SIS	7:1 0xFF62	0xFF29	1:0 0xFF11
1	SIS	SIS	0xFF29	2:0 0xFF11
2	SIS	0:2 0xFF62	0xFF29	3:0 0xFF11
3	SIS	SIS	1:2 0xFF62	4:0 0xFF11
4	SIS	2:2 0xFF62	0xFF29	0:1 0xFF11
5	SIS	SIS	0xFF29	1:1 0xFF11
6	SIS	3:2 0xFF62	0xFF29	2:1 0xFF11
7	SIS	SIS	0xFF29	0:2 0xFF11
8	SIS	4:2 0xFF62	0xFF29	1:2 0xFF11
9	SIS	SIS	0:3 0xFF62	2:2 0xFF11
10	SIS	1:3 0xFF62	0xFF29	3:2 0xFF11
11	SIS	SIS	0xFF29	4:2 0xFF11
12	SIS	2:3 0xFF62	0xFF29	5:2 0xFF11
13	SIS	SIS	0xFF29	0:3 0xFF11
14	SIS	3:3 0xFF62	0xFF29	1:3 0xFF11
15	SIS	SIS	0:0 0xFF62	2:3 0xFF11

Key	
(black)	SIS Message
(dark gray)	LLDS Port 0xFF11 Variable-Length Data Message
(light gray)	LLDS Port 0xFF29 Fixed-Length Data Message
(very light gray)	LLDS Port 0xFF62 Variable-Length Data Message

Variable-Length Data Message data is labelled X:Y, where X = MSG Frame #, Y = Sequence #

Figure 5-2: SIS/LLDS Scheduling Example – 4 PIDS PDUs/Block

NRSC-5-E Reference Document List

DHL 12/19/2022

No.	Updated for Rev. E	Doc. No.	Doc. Name	Ver.	Date	Pages in .pdf Start	End	# of pages
1	X	SY_IDD_1011s	HD Radio Air Interface Design Description Layer 1 FM - NRSC	H	November 2022	2	175	174
2		SY_IDD_1012s	HD Radio Air Interface Design Description – Layer 1 AM	G	2016-12-14	176	254	79
3	X	SY_IDD_1014s	HD Radio Air Interface Design Description Layer 2 Channel Multiplex - NRSC	K	November 2022	255	285	31
4	X	SY_IDD_1017s	HD Radio Air Interface Design Description Audio Transport	I	November 2022	286	330	45
5	X	SY_IDD_1020s	HD Radio Air Interface Design Description Station Information Service Transport	K	November 2022	331	392	62
6	X	SY_SSS_1026s	HD Radio FM Transmission System Specifications - NRSC	H	December 14, 2022	393	423	31
7		SY_IDD_1028s	HD Radio Air Interface Design Description – Main Program Service Data	E	2016-12-14	424	451	28
8	X	SY_IDD_1082s	HD Radio AM Transmission System Specifications	H	November 2022	452	493	42
9		SY_IDD_1085s	HD Radio Air Interface Design Description - Program Service Data Transport	D	2016-12-14	494	527	34
10		SY_IDD_1019s	HD Radio Air Interface Design Description – Advanced Application Services Transport	H	2016-12-14	528	559	32
11	X	SY_TN_2646s	Transmission Signal Quality Metrics for FM IBOC Signals	3	November 2022	560	616	57
12	X	SY_REF_2690s	Reference Documents for the NRSC In-Band/On-Channel Digital Radio Broadcasting Standard	3	November 2022	617	625	9
13	X	SY_IDD_4363s	HD Radio Air Interface Design Description Low-Latency Data Service Transport	A	November 2022	626	644	19