iBiquity Digital Corporation | *Technical Report*

AM Digital Data Service System Study Report

Rev. 01.14

December 15, 2010

iBiquity Digital Corporation
6711 Columbia Gateway Drive
Suite 500
Columbia, Maryland
21046
USA

443.539.4290 Telephone
443.539.4291 Facsimile

info@ibiquity.com

Table of Contents

Contents

List of Figures

List of Tables

Scope

This document summarizes the AM Digital Data Service study funded under NAB FASTROAD technology advocacy program. The AM Digital Data Service study was developed to define a data transmission service that would enable analog AM broadcasters with added capability to provide text services similar to RBDS functionality in FM. This service enhances the analog audio broadcast with a low-rate text service capability. The AM Digital Data Service is structured to be compatible with HD Radio IBOC broadcasting on AM and defined to be implemented in a cost-effective manner.

This report summarizes system design alternatives, possible transport definitions, and impacts to industry products. Several system components are borrowed from the HD Radio IBOC standard in an effort to simplify implementation. This document does not define a full standard for AM Digital Data Service nor does it define a commercial implementation. Additional system design work is required to implement and evaluate a commercial AM Digital Data Service.

Terms, Acronyms, and Abbreviations

Terms, acronyms, and abbreviations are used throughout this document and they are presented and defined in Table 1.

Table 1: Terms, Acronyms, and Abbreviations

Acronym	Definition
ADDS	AM Digital Data Service
AM	Amplitude Modulation
bps	Bits per second
DDC	Digital Down Converter
FFT	Fast Fourier Transform
FM	Frequency Modulation
IBOC	In-Band, On-Channel
IDFT	Inverse Discrete Fourier Transform
IFFT	Inverse Fast Fourier Transform
OFDM	Orthogonal Frequency Division Multiplexing
MPS	Main Program Service
PSD	Program Service Data
RBDS	Radio Broadcast Data System
SMS	Station Message Service
UI	User Interface

References

[1] iBiquity Digital Corporation
"HD Radio™ Air Interface Design Description – Layer 1 AM"
Document Number: SY_IDD_1012s

[2] iBiquity Digital Corporation
"HD Radio™ Supplemental Program Services (Multicasting)"
Document Number: RX_TN_5082

[3] iBiquity Digital Corporation
"HD Radio™ Air Interface Design Description Station Information Service Transport"
Document Number: RX_IDD_1020s

[4] iBiquity Digital Corporation
"HD Radio™ Air Interface Design Description – Program Service Data Transport"
Document Number: RX_IDD_1085s

[5] Martin Nilsson, "ID3v2.3.0 Informal standard," URL: http://www.id3.org.

[6] European Telecommunications Standards Institute (ETSI)
"Digital Radio Mondiale (DRM): AM signalling system (AMSS)"
Document Number: ETSI TS 102 386 V1.2.1 (2006-03)

1. Introduction

Analog AM broadcasters currently have no options for expanding program services to listeners. Traditional analog AM broadcast provides only audio-based services. In a multimedia age where additional services and capabilities are provided via satellite radio, HD Radio Technology, and internet connectivity, analog AM broadcasting needs an enhancement to meet the expectations of consumers. Typical consumer expectations include text information related to current programming, real-time station messages and announcements, and interactive advertisements.

The National Association of Broadcasters (NAB) is interested in proposals for expanded message service capability for analog AM broadcasters. Similar services exist for analog FM broadcasters via Radio Broadcast Data System (RBDS). With the deployment of HD Radio broadcasting in the United States, a digital data service can be created efficiently for analog AM broadcasters.

A well-designed analog AM data service can provide limited text message capability to convey information related to the current program, station information, or local issues. These services are currently part of the HD Radio system for stations that have full digital capability. A much simplified digital system can be adopted at potentially lower costs. Such a service would provide analog AM broadcasters with a step toward a full digital broadcast.

This report describes methods for digital signaling within an AM analog broadcast signal and options/tradeoffs for application layer implementation. This digital signal is intended to provide basic information about the broadcast, similar to RDS or RBDS for FM transmission. The service would support existing analog audio programming. Although other AM digital signaling methods have been proposed world-wide, the method described here is intended to be at least partially compatible with an HD Radio signal, using a subset of the modulation techniques already designed for AM HD Radio transmission equipment and receivers. The advantage is that this signaling can leverage the reuse of existing modulation and demodulation techniques in HD Radio receivers, which should facilitate adoption of the signaling method. This data service would provide broadcasters with an intermediate step toward a full digital IBOC implementation.

2. Problem Scope

2.1. Background

Transmission of data and text information on FM has been available in United States and Europe through the Radio Data System (RDS) and Radio Broadcast Data System (RBDS). In Europe, the AMSS system has been proposed for limited text messaging on AM stations [6]. Analog AM radio stations in the United States currently have no means of transmitting limited text and data services to receivers.

The IBOC modulation definitions used in HD Radio broadcasting have been proven as robust digital data carriers. These same modulation techniques may be applied on a limited scale over a few data subcarriers to enable a robust, low-rate data service. With a digital messaging capability, analog AM stations would be able to transmit information (title, artist, program messaging, and other information) related to the analog audio program.

2.2. Proposed Use Cases

The AM Digital Data Service is designed to support text messaging and data transmission associated with audio program content. Proposed use cases include station service messages, alert messages, and program service messages. These system definitions are identical to the corresponding definitions in the HD Radio system standard.

2.2.1. Station Message Service (SMS)

The station message service is intended to identify station related configuration and important messages to the listener. The station message service provides, but is not limited to, the following:

- Station Call Letters
- Station Message

This service borrows definitions from the HD Radio Station Information Service (SIS) as defined in Reference [3]. The detailed transport protocol and message definitions are provided in Section 5.2.1.

2.2.2. Program Service Data (PSD)

The program service data is intended to describe the content of the current audio program. The PSD service provides, but is not limited to, the following messages:

- Title
- Artist
- Album
- Genre
- Content
- Commercial

This service borrows definitions from the HD Radio Program Service Data (PSD) as defined in Reference [4]. The detailed transport protocol and message definitions are provided in Section 5.2.2.

2.2.3. Additional Use Cases

The AM Digital Data Service may also provide additional features such as Emergency Alert text messages or interactive advertisements. These services can be layered into the defined SMS or PSD data fields. Detailed definitions for these applications will require additional study and implementations.

2.2.4. Receiver Use Cases

A typical receiver implementing the AM Digital Data Service should have display capability to present text services to the listener. Various display types are possible and the actual layout and rendering will be dependent on the display choice. Single-line displays will limit the amount of text visible at any time. Dual-line displays should be appropriate for most use cases. High-resolution graphic displays are not required, but may be available to facilitate FM data services or other applications in the radio.

Example displays are provided below. These displays are provided as reference only. Receiver specifications will need to be developed to provide guidance to the product designers and manufacturers.

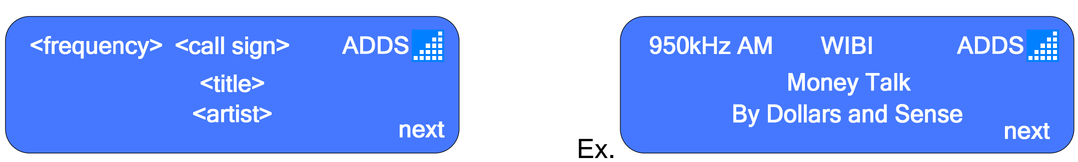


Figure 2-1: Example PSD Screen 1

Ex.

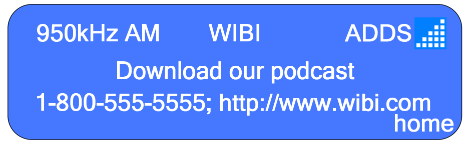


Figure 2-2: Example PSD Screen 2

Ex.

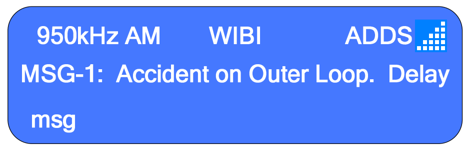


Figure 2-3: Example Message Screen

3. Waveform Definition

The AM Digital Data Service is designed to provide robust signaling capability in the AM band. This data service will be subjected to the same channel conditions and environment as analog AM broadcasts and HD Radio AM Hybrid broadcasts. These channel conditions consist of static and impulse noise, which are prevalent in the AM channel. Therefore, the data service must ensure robust digital reception in the presence of these conditions.

The ADDS system borrows heavily from the digital broadcast design defined for HD Radio AM Hybrid transmission. The design requires symbol durations, subcarrier spacing, and sample rates consistent with HD Radio broadcasts. These design requirements ensure robust reception and allow for compatibility with existing HD Radio receiver applications.

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.

The ADDS service groups OFDM symbols into logical blocks or frames. Each block consists of 32 OFDM symbols. Transmission blocks will aid in signal acquisition and data recovery.

The basic system parameters provided in Table 2 are adopted from the HD Radio AM Air Interface Definition [1].

Table 2: AM Digital Data Service System Parameters

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 Block Duration	T_b	s	$= 32\cdot T_s$	1.858×10^{-1}
L1 Block Rate	R_b	Hz	$= 1/T_b$	5.383

3.1. Subcarrier Definitions

The proposed system utilizes digital subcarriers at 181.7 Hz (±1) offset from the analog carrier, 363.4 Hz (±2) offset from the analog carrier, and (optionally) 545.1 Hz (±3) offset from the analog carrier. Each subcarrier power level is set to ensure robust reception and minimize interference to the host analog audio. A summary of the subcarrier definitions is provided in Section 3.1.2.2.

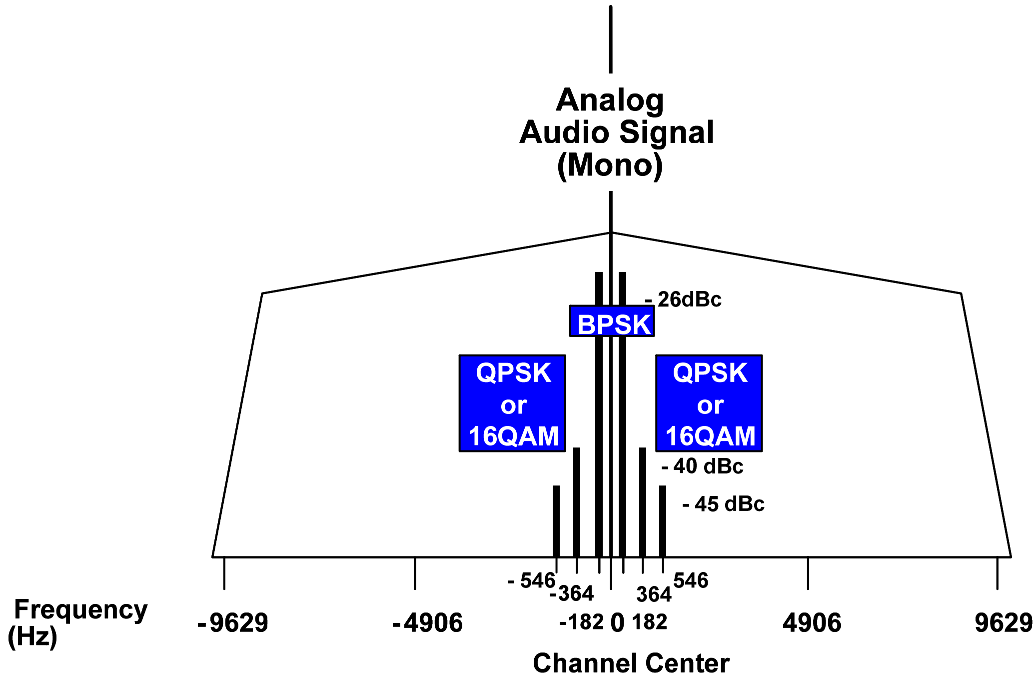


Figure 3-1: Sample Spectral Definition

3.1.1. Subcarrier ±1 Definition

It is proposed that subcarrier pair (1) be defined as the BPSK control signal of the ADDS signal. These subcarriers are modulated at a rate of approximately 172.3 Hz, and spaced approximately 181.7 Hz from the main carrier. These subcarriers must be orthogonal to the main analog carrier.

The BPSK control signal consists of a 32-bit System Control and data sequence (BPSK control signal). The system control functions are adapted from HD Radio AM transmission [1]. The 11 sync bits and 4 parity bits would remain the same as the IBOC format, although the parity would be changed from even to odd parity to prevent false detection as a valid IBOC mode (e.g., MA1) by existing HD Radio receivers. The remaining 17 bits shall be used to carry the new signaling data. These 17 bits (out of 32) would have a raw throughput of about 91.5 bps. However, the projected raw throughputs would be reduced by overhead, such as CRC.

Option: Data bit 24 can be used to indicate the first of multiple 32-bit sequences, leaving 16 bits per sequence for data. It can also indicate extension fields.

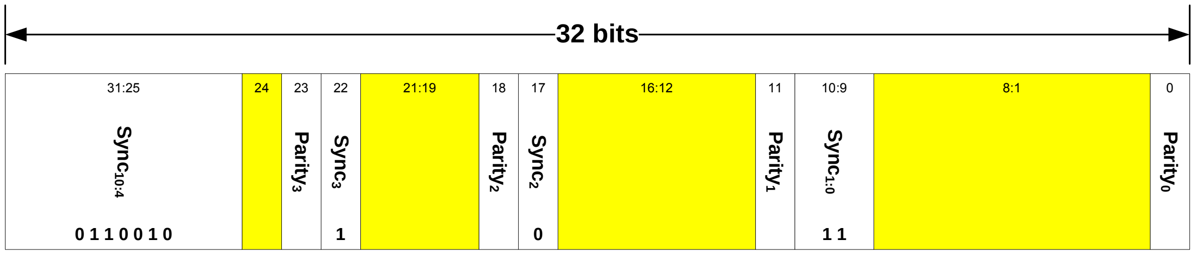


Figure 3-2: Proposed BPSK Carrier Data Pattern

Table 3: BPSK Signal Constellation Mapping

n^{th} Bit Value	n^{th} Constellation Point
0	0 – j0.5
1	0 + j0.5

3.1.2. Subcarriers ±2 and ±3 Definition

Complementary subcarriers ±2 and ±3 may be optionally added to the AM Digital Data Service to provide increased capacity. These subcarriers would provide additional capacity for expanded text services. There are two modulation options possible for these added carriers.

The same carrier synchronization, symbol synchronization, and framing synchronization algorithms used in present IBOC receivers can be reused for this application. However, it is not necessary to use the training symbols for QPSK equalization. This is because subcarriers ±2 and ±3 are close in frequency (correlated) to the main carrier (phase). Branch metrics for soft convolutional decoding can be derived after flat-fade equalization (existing receivers), estimating the QPSK amplitudes, and then estimating the noise from the nominal QPSK (QAM) constellation points. Reed-Solomon coding is an alternative method that can avoid the need for soft decoding branch metrics.

3.1.2.1. QPSK Modulation Option

These subcarriers can use QPSK modulation, but should be transmitted at a lower power than the BPSK subcarrier pair. Consequently the coverage of these additional subcarriers is not as robust as the BPSK subcarriers. Each additional pair of complementary subcarriers would add a raw throughput of about 344.5 bps. Forward Error Correction coding for these subcarriers is recommended which will reduce the overall throughput.

Table 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

3.1.2.2. QAM Modulation Option

Alternatively, the subcarrier pairs (2) and (3) may be modulated using 16-QAM. This modulation will increase the data capacity on the subcarriers. However, there will be an impact on robustness and receiver complexity. These carriers should be transmitted at a lower power than the BPSK subcarrier pair. Consequently the coverage of these additional subcarriers is not as robust as the BPSK subcarriers. Additional channel state estimation will be required in the receiver to properly track and decode the data carriers. Each additional pair of complementary subcarriers would add a raw throughput of about 689 bps. Forward Error Correction coding for these subcarriers is recommended which will reduce the overall throughput.

Table 5: 16-QAM Constellation Mapping

2 bit word x_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

3.1.3. Modulation Summary

The proposed system modulation could be expandable based on each individual station need for data throughput and services. The summary capacity table is presented in Table 6. The maximum coded throughput in this proposed system would be 1098 bits per second.

Table 6: Suggested Subcarrier Levels and Data Throughput

Carrier/Modulation	Power Level (dBc)	Raw throughput (bps)	Coded Throughput (Convolutional)	Coded Throughput (Golay)	Coded Throughput (Reed-Solomon)
± 1 : BPSK	-26	91.5			
± 2 : QPSK	-40	344.5	230 bps	172 bps	258 bps
± 3 : QPSK	-44.5	344.5	230 bps	172 bps	258 bps
± 2 : 16-QAM	-40	689	460 bps	345 bps	517 bps
± 3 : 16-QAM	-44.5	689	460 bps	345 bps	517 bps

3.2. Forward Error Correction Coding Analysis

Forward Error Correction (FEC) Coding is recommended for the data transmitted on complementary subcarrier pairs ±2 and ±3. FEC coding is not recommended for the BPSK complementary subcarriers pair ±1 for several reasons. Subcarrier pair ±1 is transmitted at a higher level, BPSK is more robust than other modulation types, the throughput is low (91.5 bps) and FEC would reduce the effective throughput, and the data on the BPSK subcarriers is transmitted frequently and is already parity-checked for errors.

The FEC codes are considered here:

1) Convolutional code (e.g. K=9, tailbiting);

2) Golay code, extended (24,12); and

3) Reed-Solomon block code [e.g., modified RS(16,12) over GF 2^4].

The pros and cons of each are considered.

The tailbiting, K=9, convolutional code is already implemented in the IBOC system for short messages. The code rate implemented in the IBOC PIDs information messages is R=2/3, meaning that the ratio of information bits to coded bits is 2/3. Although convolutional codes are more-frequently used for semi-infinite data streams, or used with K-1 (e.g., 8) flush bits for moderate size messages. When the messages are very short, the flush bits consume throughput capacity. Tailbiting is a technique that eliminates the need for flush bits, and the convolutional code is effectively used as a block code. However, the error-correcting ability of the convolutional code may be reduced when the messages are very short. Furthermore, convolutional codes exploit soft input information about the quality of each symbol, weighting the corrupted or noisy symbols less than the more-reliable symbols, if that can be assessed. This soft-decision advantage is effective for AWGN channels; however, soft-decision estimation of complex channel gain and noise variance estimates are needed; these are partial byproducts of equalization. The AM band noise may be characterized as more impulsive than AWGN, so the soft-decision advantage may not be much better than erasure-decoding used on other block codes.

The extended (24,12) Golay code transforms 12 information bits into a 24-bit codewords (R=1/2). This is a block code that can correct up to 3 bit errors in each block. This code is optimal in the sense that there can be no better code of this size (23,12) that can correct more than three errors. However, larger codes can be more efficient. This code is most useful if messages sizes are multiples of 24 bits. The (23,12) Golay code can be considered if this better-fits the message framing.

The Reed-Solomon (RS) block code can be a good choice for messages of similar size as the codewords. Furthermore RS codes are suitable for impulsive-like noise, such as characterized in the AM band. The RS code is very efficient in terms of error-correcting efficiency and associated overhead. It is optimal in the sense that it requires the fewest number of parity-check symbols for the number of corrected symbol errors. The systematic property of RS codes uses the un-coded message bits (symbols), appended with RS parity symbols for error correcting. So it is possible (e.g., for inexpensive receivers) to directly read the message without decoding, but with no error correcting ability. Specifically the R=3/4, modified RS(16,12) [optionally RS(15,11) if framing is more convenient] over GF 2^4 is recommended for this short message application. The 64-bit codewords consist of 16, 4-bit symbols of which 12 symbols (48 bits) are information. This code has the ability to correct up to two symbol errors (each four bits) per codeword. Up to four symbol errors per codeword can be corrected with erasure decoding. Because of the bit-grouping in symbols, this code is well-suited for burst errors, or errors that can affect whole symbols. For example, impulsive noise with 16-QAM symbols would generally have burst errors of four bits. Also two successive QPSK symbols may be affected by impulsive noise.

The RS(16,12) code over GF 2^4 is tentatively recommended for this application. The 64-bit codewords contain 48 information bits. The 64-bit codewords contain 48 information bits. Although this code reduced the throughput to R= 3/4, its performance permits roughly a 6-dB reduction in CNR at the same BER.

3.3. OFDM Generation

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 Digital Data Service signal. A conceptual block diagram of OFDM signal generation is shown in Figure 3-3.

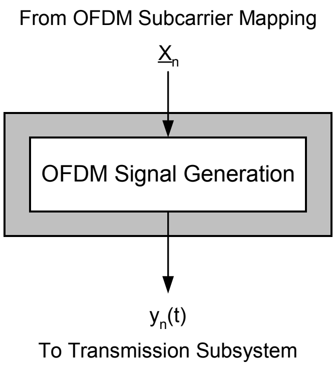


Figure 3-3: OFDM Signal Generation Conceptual Block Diagram

The input to OFDM signal generation is a complex vector, X_n of length L, representing the complex constellation values for each OFDM subcarrier in OFDM symbol n.

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.

Let $X_n[k]$ be the complex scaled constellation points from OFDM subcarrier mapping for the nth symbol, where k = 0, 1, …, L-1 indexes the OFDM subcarriers. Let $y_n(t)$ denote the complex time-domain output of OFDM signal generation for the nth symbol. Then $y_n(t)$ can be written in terms of $X_n[k]$ as follows:

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

where n = 0,1,…,∞, 0 ≤ t ≤ ∞, L = 163 is the maximum number of OFDM subcarriers, and T_S and Δf are the OFDM symbol period and OFDM subcarrier spacing, respectively.

The signals can be generated (and received) at the existing complex baseband AM sample rate (46,511.781875 complex samples per second). This sample rate can be conveniently derived from a 10 MHz oscillator. Division of 10 MHz by 215 yields 46,511.628 Hz (about 3.3 ppm error). The transmitter performs modulates OFDM symbols with a 256-sample complex IFFT, although computing the IDFT over a small number of subcarriers could be more efficient than the FFT.

3.4. Symbol Generation

The OFDM symbols are appended with a 14-sample cyclic prefix, and tapered with a root raised cosine pulse shape over the 14 symbols at each end. The cyclic prefix results in 270-sample symbol (5.8 msec). Although the pulse shape is convolved with an additional Gaussian-shaped pulse, it is not necessary for this AM signaling system.

$$H(\xi)=\begin{cases} 0.5\cdot\left[1+\cos\left(\pi\dfrac{\alpha T-\xi}{\alpha T}\right)\right], & \text{for } 0<\xi\leq\alpha T \\ 1.0, & \text{for } \alpha T<\xi<T \\ 0.5\cdot\left[1+\cos\left(\pi\dfrac{\xi-T}{\alpha T}\right)\right], & \text{for } T\leq\xi\leq(1+\alpha)T \\ 0, & \text{otherwise} \end{cases}$$

α is the cyclic prefix width defined in Table 2 and T = 1/Δf is the reciprocal of the OFDM subcarrier spacing. Figure 3-4 shows a plot of the pulse shaping function W(ξ).

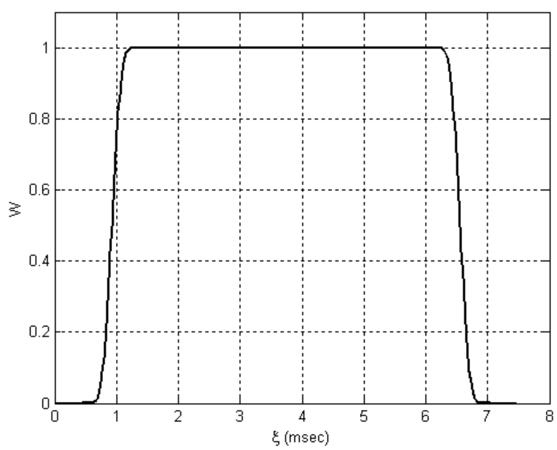


Figure 3-4: Pulse Shaping Function

3.5. Analog AM Modulator

When broadcasting the ADDS waveform (analog carrier combined with OFDM subcarriers), this process computes the envelope of the analog AM Digital Data Service 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 Exciter, so that a(t) does not assume negative values.

3.6. Analog/Digital Combiner

When broadcasting the ADDS 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 waveform z(t), as follows:

Re [z(t)] = Re [y(t)] + a(t)

Im [z(t)] = Im [y(t)]

4. Performance Study

4.1. Subcarrier Impact to Analog

The proposed modulation for AM Digital Data Service will require digital subcarriers transmitted under the analog modulation. These subcarriers have the potential to generate noise on certain receivers tuned to the analog broadcast. The impact of subcarriers ±1, ±2 and ±3 under the analog audio was analyzed.

Because the digital subcarriers are transmitted in quadrature (complementary subcarrier pairs) to the DSB analog audio signal, their effect on coherent AM detectors is theoretically null. Although the group delay variation over the BPSK signal bandwidth due to imperfections in the analog IF filter would result in digital-to-analog crosstalk, this should be very small over the narrow bandwidth encompassing the BPSK signal (subcarriers ±1). A coherent AM detector will outperform the envelope detector.

For envelope detectors, the crosstalk is suppressed by approximately double dB relative to the main carrier level. Double dB concept with quadrature signal on envelope detectors:

Assume we have a main carrier, unmodulated with a normalized level of a constant 1. Now suppose we modulate the carrier with normal AM modulation with a small noise x<<1. A coherent or envelope detector would subtract the mean (1) from the instantaneous signal (1+x) to get the output result x.

Then the noise power relative to the main carrier level in dBc is computed as $10 \cdot \log\left(x^2\right)$ dBc

Now suppose this same noise x is modulated in quadrature to the carrier ($1 + j \cdot x$), and detected with an envelope detector with output $\sqrt{1+x^2} - 1$. For small x, we can expand into a power series to approximate the result.

$$\sqrt{1+x^2} - 1 \cong \left(1 + \frac{x^4}{2}\right) - 1 = \frac{x^4}{2}$$

Then the noise power relative to the main carrier level in dBc is computed as:

$$10 \cdot \log\left(\frac{x^4}{4}\right) = 2 \cdot 10 \cdot \log\left(x^2\right) \text{- } 6 \ \text{dBc}$$

So we have shown that the quadrature noise is suppressed by twice the dBc ratio plus another 6 dB, relative to the in-phase modulation (signal).

For example, if the BPSK subcarrier pair is -23 dBc, then the crosstalk into an envelope detector is about -52 dBc (2 × 23+6) according to the analysis above. Since typical AM audio processors maintain the analog modulation at approximately -13 dBc, the resulting SNR is -39 dBc. Although this SNR may not seem great, its effect is further reduced by the low-frequency cutoff of the audio bandpass filter, and substantially suppressed by the human ear’s sensitivity to low frequencies (A-weighting, or ITU-R 468 weighting). The plot in Figure 4-1 shows that the sensitivity to BPSK noise centered at subcarriers ±1 (approximately 182 Hz) is roughly 15 lower than the normalized sensitivity at 1 kHz. Therefore the effective (weighted) SNR due to the BPSK signal in a receiver with an AM envelope detector is roughly 54 dB. This could be degraded further by IF filter asymmetry, mistuning, and high levels of modulation making the “double-dB” approximation less accurate. Actual receivers should be evaluated for effective SNR performance with the BPSK subcarriers. The additional effects of adding subcarriers ±2 and ±3 are analyzed next.

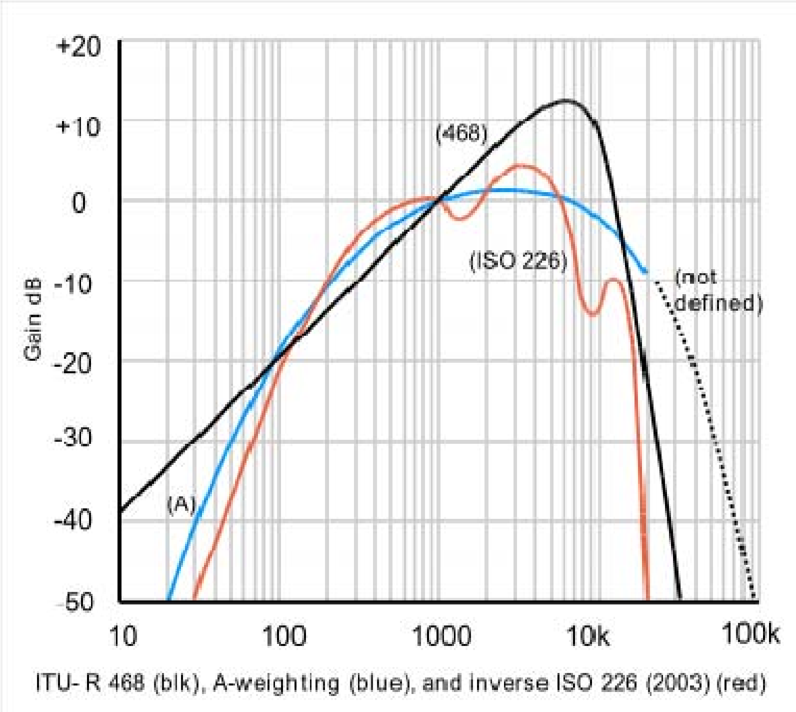


Figure 4-1: Relative Sensitivity of the Human Ear to Sounds at Various Frequencies

The SNR degradation due to the addition of subcarriers ±2 and ±3 is analyzed by adding these subcarriers at levels resulting in 1 to 3 dB degradation in audio quality relative to the BPSK subcarrier pair ±1 alone. Of course it is also possible to reduce the entire set of subcarriers to any level relative to the BPSK SNR performance. The human ear sensitivity at the subcarrier frequencies is also taken into consideration. The sensitivity plot shows that sensitivity falls by 20 dB per decade at lower frequencies in the subcarrier frequency range. This is equal to six dB per octave. Since subcarrier pair ±2 is an octave above subcarrier pair ±1, then the ear is six dB more sensitive to subcarrier pair ±2 at the same level. The ear is 9.54 dB more sensitive at subcarrier ±3 frequencies than subcarrier ±1. Therefore the levels of subcarriers ±2 and ±3 should be further suppressed by six and 9.54 dB after the objective levels are computed for 1-dB and 3-dB SNR degradation. The additional audio SNR degradation for adding subcarriers ±2 and ±3 can be expressed as

$$SNRdegradationdB = 10 \cdot \log\left[\frac{10^{SC1dBc/10} + 10^{(SC2dBc+6)/10} + 10^{(SC3dBc+9.54)/10}}{10^{-26/10}}\right],$$

where $SC1dBc$, $SC2dBc$, and $SC3dBc$ are the subcarrier (±1, ±2, and ±3) levels in dB. Below is Table 7 of subcarrier levels versus effective SNR performance after the sensitivity adjustments are accommodated. The highlighted rows are the cases plotted in the BER performance section.

Table 7: Subcarrier Levels vs. Audio SNR Degradation Relative to Reference BPSK Level

BPSK, *SC1dBc*, dBc	Subcarriers *SC2dBc*, dBc	Subcarriers *SC3dBc*, dBc	Weighted Audio SNR Degradation, dB
-26	X	X	0 dB, reference
-27	-39	X	0
-27	-41	-45.5	0
-27	-44	-44	0
-29	-37	-43	0
-29	-40	-40	0
-26	-38	X	1
-26	-40	-44.5	1
-26	-43	-43	1
-26	-32	X	3
-26	-34	-40	3
-26	-37	-37	3

4.2. Analysis of Performance and Coverage

The coverage of the digital signal depends on its signal strength and the receiver noise or ambient noise level. The receiver noise of good AM tuners, such as car tuners, is generally lower than the ambient noise. However, most portable and tabletop AM receivers are limited by receiver noise. So absolute coverage is a function of the signal strength, and is also receiver-dependent. A more meaningful and useful method of characterizing coverage is to relate the corresponding analog audio quality to the digital reception. Particularly, for a given carrier to noise density ratio C/No, the BER of the digital signal corresponds to an audio SNR of the analog AM host signal. This relationship can be used to determine to what extent the digital signal is available over the useful listening area.

The probability of bit error, or BER, of a BPSK signal can be computed as a function of its energy-per-bit Eb (joules) and the noise density No (W/Hz). Additive White Gaussian Noise (AWGN) is assumed in the computation.

$$BER = \frac{1}{2} \cdot erfc\left(\sqrt{\frac{Eb}{No}}\right)$$

Since we want to solve for BER as a function of C/No, then we must relate the value of Eb to C. Assuming each BPSK subcarrier level is SC1dBc (e.g., -26 dBc), then the complementary pair results in twice as much bit energy

$$Eb_{BPSK} = C \cdot T \cdot 2 \cdot 10^{SC1dBc/10}$$

where T is the reciprocal of the bit rate, or approximately 0.0058 seconds in this case. Then the BER can now be expressed as a function of C/No. For convenience, C/No is expressed in dB-Hz as CNR.

$$\boxed{BER_{BPSK} = \frac{1}{2} \cdot erfc\left(\sqrt{2 \cdot T} \cdot 10^{(CNR+SC1dBc)/20}\right)}$$

The BER of the QPSK is simply related to that of the BPSK signal. An offset of three dB is incurred since two bits are transmitted per symbol, requiring twice the power of BPSK for the same BER.

$$Eb_{QPSK} = C \cdot T \cdot 10^{SC1dBc/10}$$

$$\boxed{BER_{QPSK} = \frac{1}{2} \cdot erfc\left(\sqrt{T} \cdot 10^{(CNR+SC2dBc)/20}\right)}$$

The BER performance of 16-QAM can be analyzed (approximated) in a similar way.

$$BER_{16QAM} \cong \frac{3}{8} \cdot erfc\left(\sqrt{\frac{2}{5} \cdot \frac{Eb}{No}}\right)$$

$$Eb_{16QAM} = \frac{1}{2} \cdot C \cdot T \cdot 10^{SC1dBc/10}$$

$$\boxed{BER_{16QAM} \cong \frac{3}{8} \cdot erfc\left(\sqrt{\frac{T}{5}} \cdot 10^{(CNR+SC2dBc)/20}\right)}$$

The plots of Figure 4-2 show the relative performance of the BPSK, QPSK and 16-QAM subcarriers, all at -26 dBc. QPSK and 16-QAM incur approximately a 3-dB and a 10-dB CNR penalty, respectively.

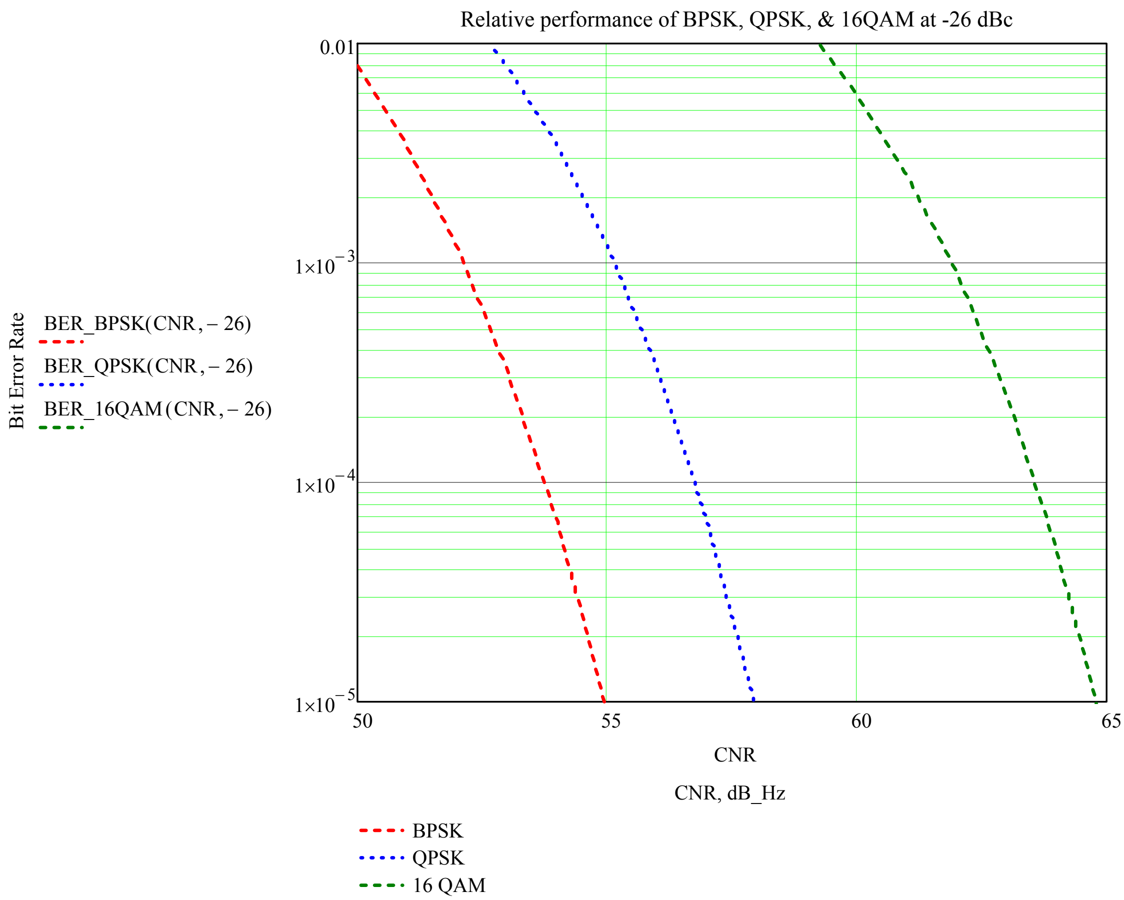


Figure 4-2: Relative BER Performance of BPSK, QPSK, and 16QAM at -26dBc

The analog audio SNR (in dB) can also be expressed as a function of CNR. The audio SNR is characterized in a 5-kHz audio bandwidth. No adjustment (such as A-weighting) is assumed here. It will be assumed that a coherent AM detector is used (instead of an envelope detector), to simplify the calculation. Although the DSB signal extends over 10 kHz (±5 kHz), the DSB demodulation effectively sees the signal and noise No over five kHz. Assume that typical AM audio processors set the average analog signal level to -13 dBc:

$$SNR = CNR - 13 - 10 \cdot \log(5000)$$

or

$$\boxed{SNR = CNR - 50}$$

The plots of Figure 4-3 to Figure 4-6 show the digital and analog performance versus CNR.

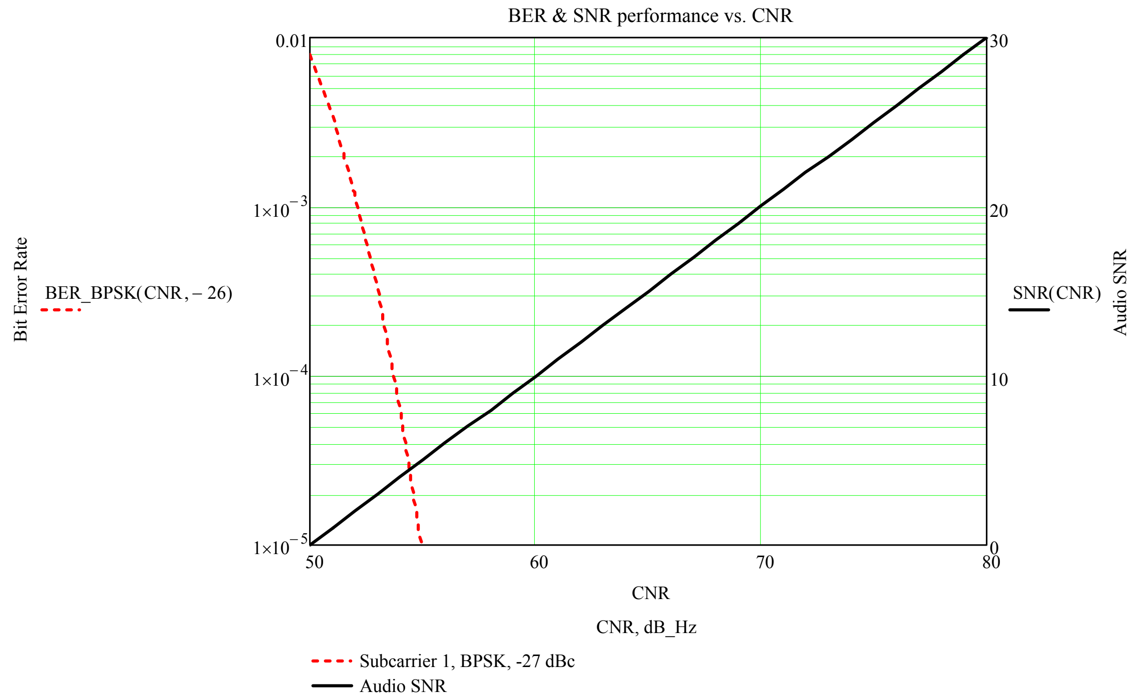


Figure 4-3: Digital and Analog Audio Performance vs. CNR. Single BPSK subcarrier

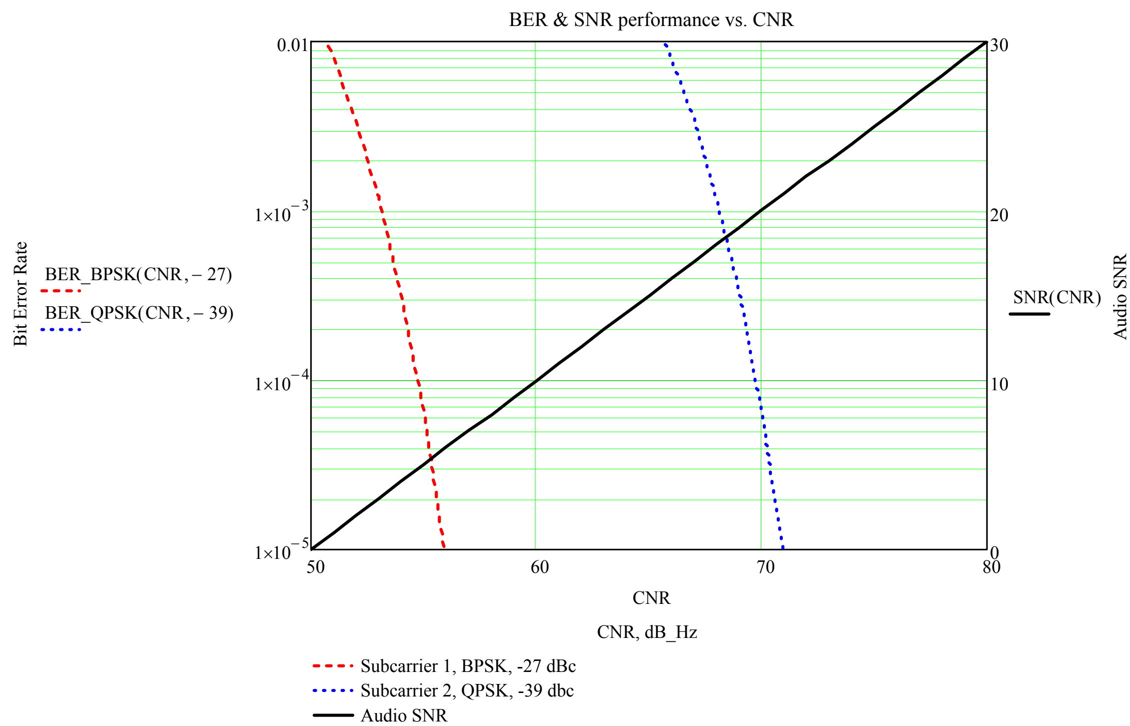


Figure 4-4: Digital and Analog Audio Performance vs. CNR. Additional QPSK subcarrier pair ±2. Levels adjusted for no analog SNR degradation.

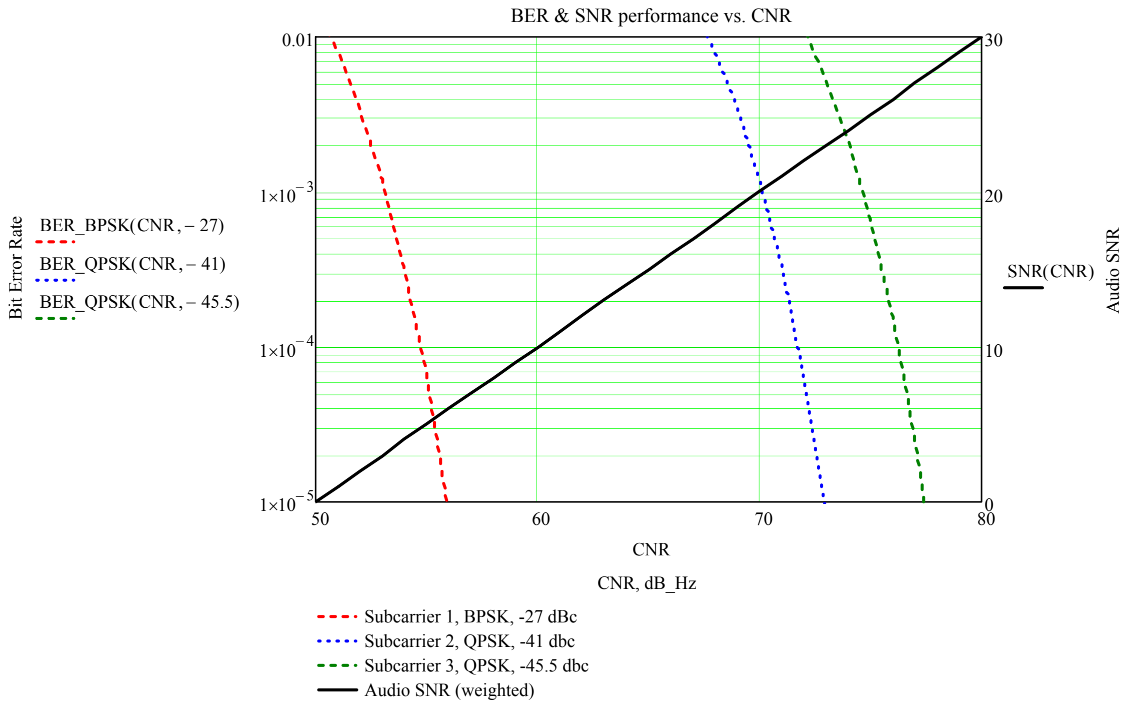


Figure 4-5: Digital and Analog Audio Performance vs. CNR. Two additional QPSK subcarrier pairs. Levels adjusted for no analog SNR degradation.

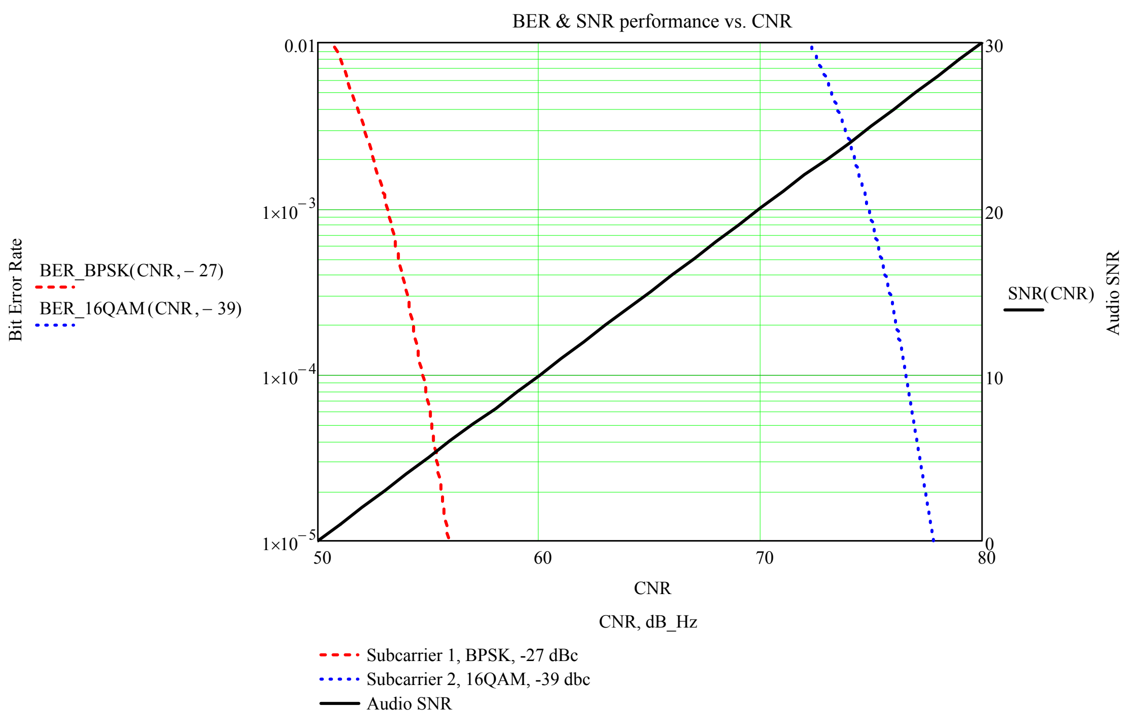


Figure 4-6: Digital and Analog Audio Performance vs. CNR. Additional 16QAM subcarrier pair ±2. Levels adjusted for no analog SNR degradation.

An analog audio SNR of 26 dB is typically used as a threshold for AM receiver sensitivity. It is remarkable that the digital signal is uncorrupted at this corresponding CNR=76 dB-Hz. The BPSK signal is somewhat useful for intermittent messaging (BER=0.01) where the audio SNR is zero dB. Similarly the QPSK signal is useful where the audio SNR is 10 dB. FEC coding could make the digital signals more robust, at the expense of some throughput.

The signal field strength for a given CNR can be computed if the ambient noise level is known. The antenna noise factor Fa is the excess noise above the thermal noise (kT). The field strength of the noise in a 1-Hz bandwidth can be expressed as

$$En = Fa + 20 \cdot \log\left(f_{MHz}\right) - 95.5 \ \ \mathrm{dBu}/\sqrt{\mathrm{Hz}}$$

A value of Fa=55.5 dB is in the typical (but possibly quiet) range. At 1-MHz AM carrier frequency, the corresponding Noise density field strength is

$$En = -40 \ \ \mathrm{dBu}/\sqrt{\mathrm{Hz}}$$

Then the signal strength in dBu is related to CNR (dB-Hz) by subtracting 40 dB, for this example. The plot of Figure 4-7 is modified by replacing the CNR with signal field strength E dBu. It is assumed that the receiver noise is negligible compared to the ambient noise.

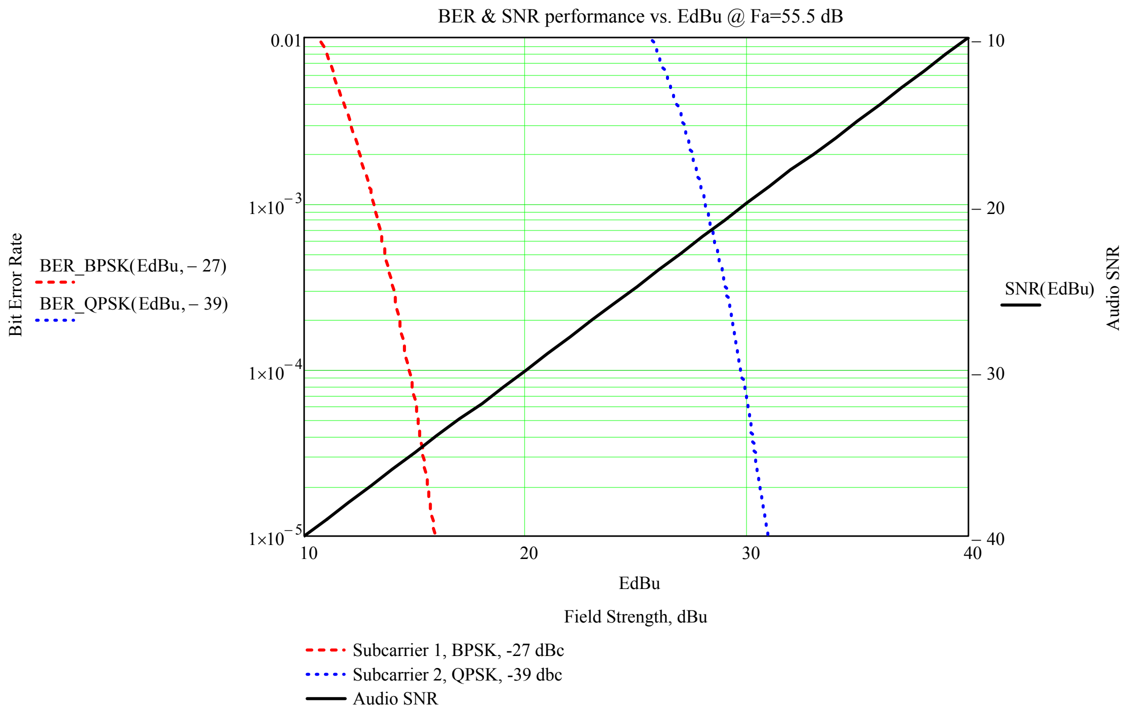


Figure 4-7: Digital and Analog Audio Performance versus Signal dBu at Fa = 55.5 dB

5. Transport Layers

Selecting the appropriate transport layer definition will be dependent on the desired use cases and selected modulation schemes. Specific transport layers work best with certain subcarrier data definitions.

A number of factors will impact the choice of protocols used. The system must account for broadcaster use scenarios, desired data throughputs, and receiver acquisition times. Additionally, system performance tradeoffs may benefit from existing definitions and protocol implementations already defined in the HD Radio AM transmission standard.

5.1. Use Case Options

This section discusses possible modulation options to enable transport of the service information proposed in Section 2.2. The options listed below define possibilities for mapping different service features based on available data bandwidth and service priorities. It is planned that the ADDS service options can be scalable or extensible based on the broadcasters' desired operation plan.

5.1.1. Protocol and Subcarrier Mapping Option 1

This option is defined for a rapid acquisition of station service information. This configuration would be beneficial to optimize tuning times and rapid acquisition of station content configuration. In this use case definition, the Station Information Service (SIS) defined for HD Radio AM broadcasting would be directly implemented on subcarrier pair (2) with a high throughput.

Suggested implementation:

1. Subcarrier ±1; BPSK: Provides 'Core PSD' including artist, title, genre, call sign. All provided in ID3/PSD tags. Transport will be minimal relying on natural 32-symbol framing and parity of the BPSK sequence. See Section 5.2.3 for PSD transport definition and Section 6.1.3 for PSD message fields.
2. Subcarrier ±2; 16-QAM: Provides SIS 'services' message for scanning for alert providers, provides emergency alerts when in effect, provides station message. Supports rapid scan and station identification. See Section 5.2.2 for SIS transport definition.
3. Subcarrier ±3; 16-QAM: This subcarrier may be added optionally at lower level, providing 'supplemental PSD'. The supplemental fields may include longer PSD messages for comments and commercials. See Section 5.2.3 for PSD transport definition and Section 6.1.3 for PSD message fields.

5.1.2. Protocol and Subcarrier Mapping Option 2

This option is defined for transport of minimal station service information and support for longer PSD data transport. This configuration would be beneficial to optimize tuning times and rapid acquisition of station content configuration. In this use case definition, the Station Information Service (SIS) defined for HD Radio AM broadcasting would be directly implemented on subcarrier pair (2) with a high throughput.

Suggested implementation:

1. Subcarrier ±1; BPSK: Provides basic station message service with minimal content for identification. See Section 5.2.1 for SMS transport definition.
2. Subcarrier ±2; QPSK: This subcarrier may be added optionally providing core PSD data fields. Long messages for program content (title, artist, and album or show name). See Section 5.2.3 for PSD transport definition and Section 6.1.3 for PSD message fields.
3. Subcarrier ±3; QPSK: This subcarrier may be added optionally at lower level, providing 'supplemental PSD'. The supplemental fields may include longer PSD messages for comments

and commercials. See Section 5.2.3 for PSD transport definition and Section 6.1.3 for PSD message fields.

5.2. Transport Definitions

5.2.1. Station Message Service Transport Definition

The Station Message Service (SMS) provides broadcast station identification and control information. SMS is transmitted in a series of SMS Protocol Data Units (PDUs) on the Primary IBOC Data Service (PIDS) logical channel. SMS Short PDUs are 34 bits in length (22-bit message payload) as shown in Figure 5-1 or 68 bits in length (48-bit message payload) as shown in Figure 5-2. 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 normally set to zero (0) for a short payload format transmitted over a 2-block period (371.5ms). The Type bit is set to one (1) for a long payload format transmitted over a 4-block period (743 ms). The Station Message Service transport definition proposed for ADDS is derived from the corresponding SIS definition used in the HD Radio system [3][4].

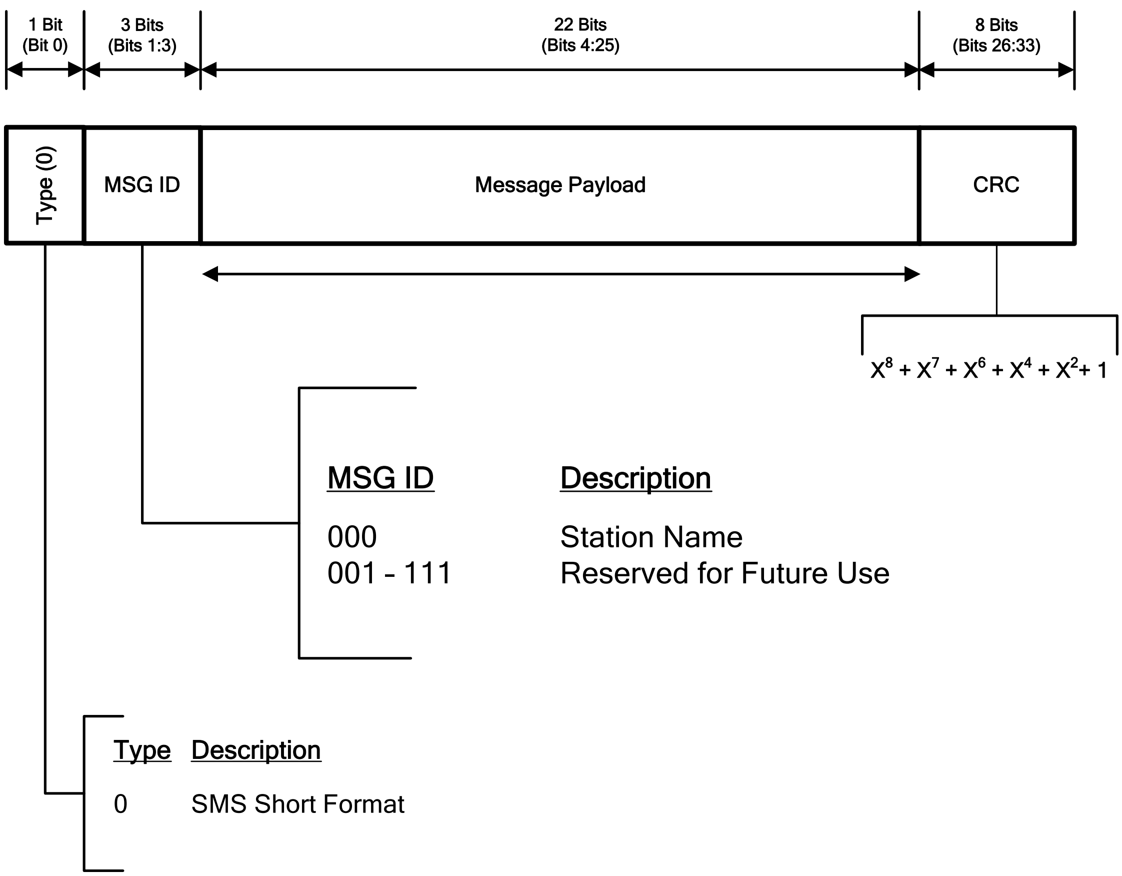


Figure 5-1: SMS PDU Format – Type = 0

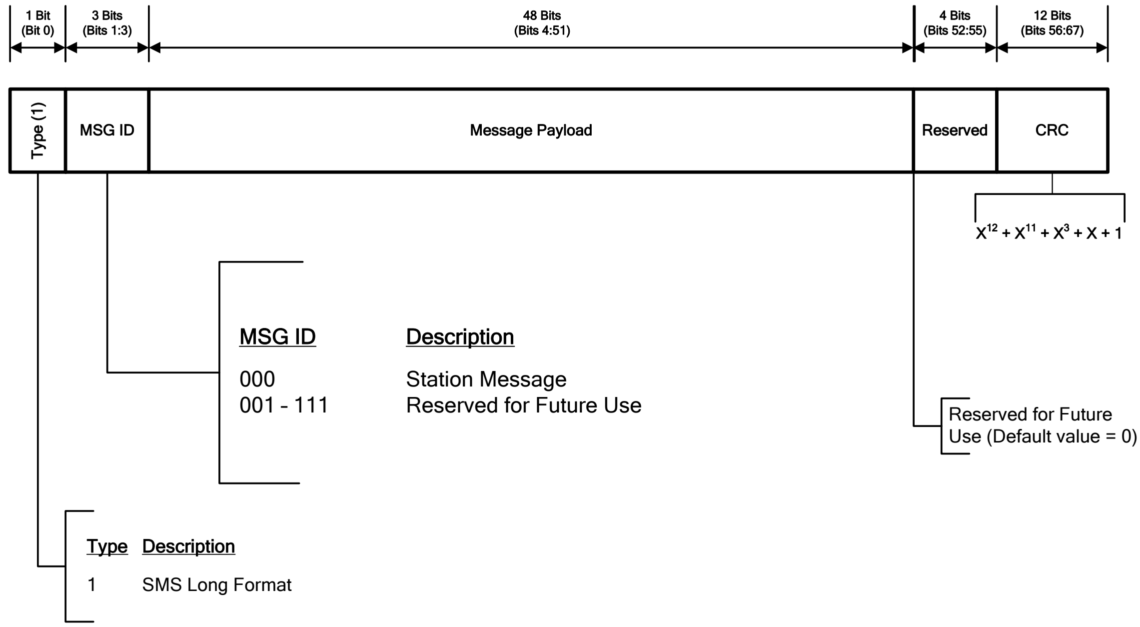


Figure 5-2: SMS PDU Format – Type = 1

5.2.1.1. Station Name – Short Format (MSG ID = 0000)

This message type has both a short format and a long format. The short format allows 4-character call signs. A long format (up to 8 characters) can be generated by defining the extension bit and transmitting an additional 4-character Station Name field. The Station Name definition proposed for ADDS is derived from the corresponding SIS definition used in the HD Radio system [3][4].

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 5-3 shows the message structure for the Station Name (short format).

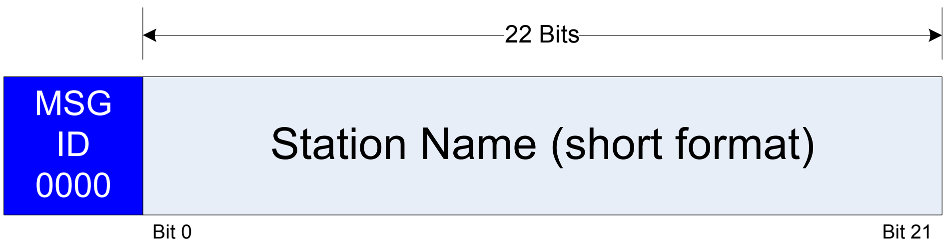


Figure 5-3: Station Name (short format) – Message Structure

Each character is five bits in length (MSB first, or leftmost), followed by a 1-bit extension. Refer to Table 8 for details of the field bit assignments (positions) and Table 9 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 1-bit extension may be used to append additional four characters.

Table 8: 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: 1 = Reserved for future use

Table 9: Station Name (short format) – Character Definitions

Value (MSB:LSB)	Character
00000, 00001, 00010,…, 11001	A, B, C,…, Z
11010	space character
11011	?
11100	-
11101	*
11110	$
11111	Reserved

5.2.1.2. Station Message (MSG ID 1000)

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 48 bits. This message can span over multiple frames. The Station Message definition proposed for ADDS is identical to the corresponding SIS definition used in the HD Radio system [3][4]. Figure 5-4 shows the message structure for the Station Message. The format of the first frame is different from the others, as shown.

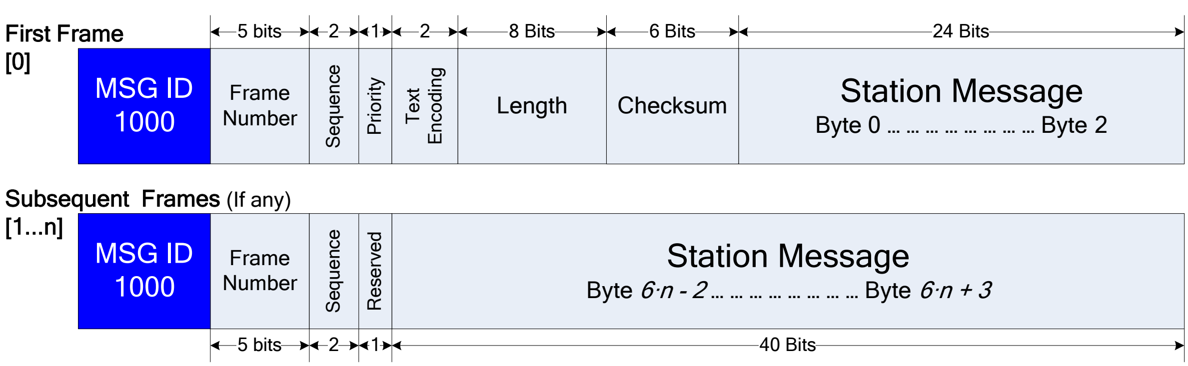


Figure 5-4: Station Message – Message Structure

The Station Message can be used to send a string of up to 158 8-bit characters or 79 16-bit characters 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 6-bit checksum is included in the first frame to increase receive reliability.

Table 10 and Table 11 describe the data fields for the first and subsequent frames, respectively.

Table 10: Description of Station Message Fields for Frame Number = 0

Field Name	Range	Description
Frame Number	0 - 31	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	N/A	See Table 12
Length	4 - 158	Defines the total number of bytes of the Station Message text, excluding any unused bytes in the last frame
Checksum	0 - 63	The modulo-64 sum of all the data bytes of the Station Message text, excluding overhead bytes
Station Message	N/A	For 8-bit character encoding, frame 0 contains the first 3 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 11: Description of Station Message Fields for Frame Number = 1 to n

Field Name	Range	Description
Frame Number	0 - 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 – 1	Reserved for future use

Field Name	Range	Description
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)

Table 12: Text Encoding Definitions

Value	Service Type
00 (default)	ISO/IEC 8859-1:1998
01	Reserved
10	ISO/IEC 10646-1:2000 UCS-2 (Little Endian)
11	Reserved

5.2.1.3. CRC Field – Short Format (Type = 0)

Each PDU is terminated with an 8-bit Cyclic Redundancy Check (CRC) for the purpose of aiding the receiver in detecting transmission errors. The CRC, ordered as PDU bits 26:33, is computed as follows:

1. Fill PDU bits 26:33 with zeros.
2. Perform modulo-two division of PDU bits 33:0 by the generator polynomial g(x),

   Where $g(x) = X^8 + X^7 + X^6 + X^4 + X^2 + 1$

   and PDU bit 33 is computed first.
3. The 8-bit remainder is then copied back into PDU bits 26:33, where bit 26 is considered the most significant remainder bit and bit 33 is the least significant remainder bit.

5.2.1.4. CRC Field – Long Format (Type = 1)

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 56:67, is computed as follows:

1. Fill PDU bits 56:67 with zeros.
2. Perform modulo-two division of PDU bits 67:0 by the generator polynomial g(x),

   Where $g(x) = X^{12} + X^{11} + X^3 + X + 1$

   and PDU bit 67 is computed first.
3. The 12-bit remainder is then copied back into PDU bits 56:67, where bit 56 is considered the most significant remainder bit and bit 67 is the least significant remainder bit.

5.2.2. Station Information Service Transport Definition

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 16-QAM modulated subcarrier (2). SIS PDUs are 80 bits in length as shown in Figure 5-5. 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 normally set to zero. If this bit is a one, the remainder of the PDU contents may be different. This option is reserved for future use. The Station Information Transport definition proposed for ADDS is identical to the corresponding SIS definition used in the HD Radio system [3][4].

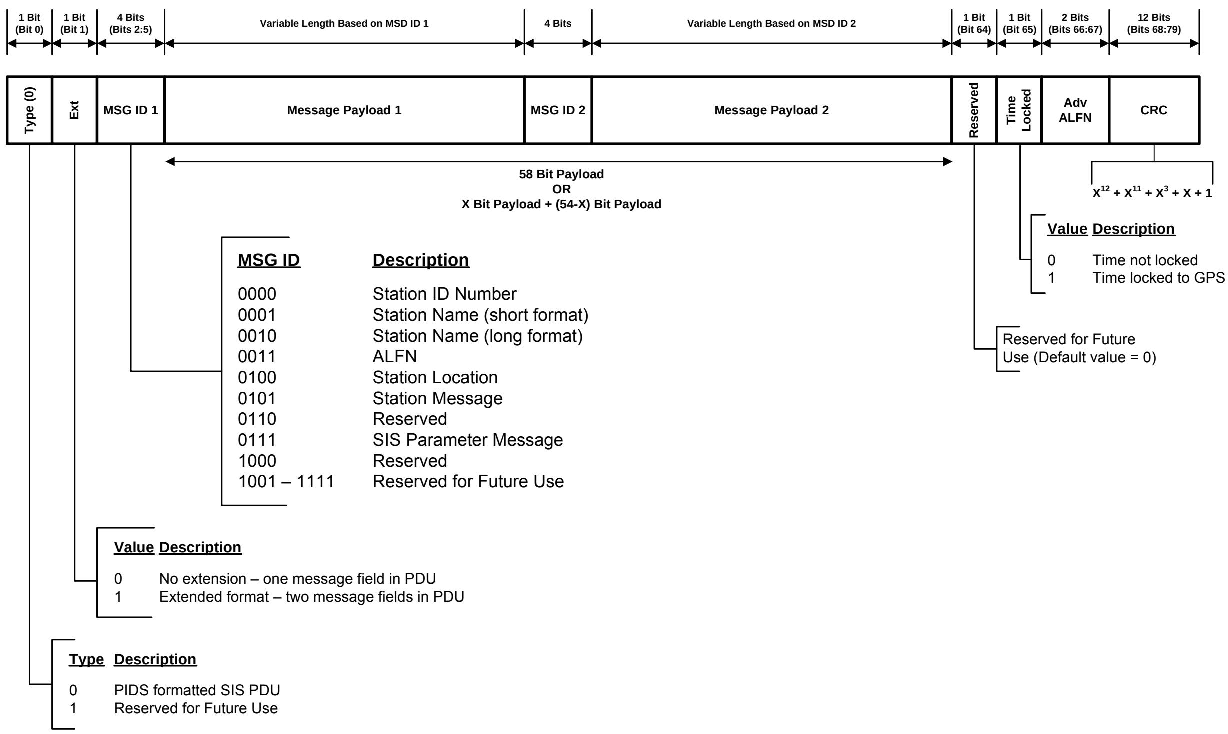


Figure 5-5: SIS PDU Format – Type = 0

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 1 field is up to 58 bits in length and the message contents are determined by the state of the first message ID field, MSG ID 1. Any unused bits at the end of the message payload 1 field are zeroed. If the Ext bit equals 1, then the first message has its length and contents defined by MSG ID 1, and the second message is active, with length and contents defined by MSG ID 2. In this case, the combined lengths of the two messages must be no greater than 54 bits. Any unused bits at the end of message payload 2 are zeroed.

The definitions of the MSG ID 1 and MSG ID 2 fields are identical. Refer to [3] for details of the MSG ID field. Any message may be placed in either message 1 or message 2 provided that the total 56-bit available payload length is not violated. Longer messages must use the single message option (Ext = 0).

Table 13: 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
0011	32	ALFN	Identifies the current Absolute Layer 1 Frame Number (ALFN)
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	Reserved	Reserved
0111	22	SIS Parameter Message	Carries the Leap Second/Time Offset and Local Time data parameters
1000	58	Reserved	Reserved
1001 - 1111	–	Reserved	Reserved for future use

The details of the SIS transport protocol and message definitions are provided in Reference [3]. The details of the message transport will not be repeated for this report.

5.2.3. PSD Transport

The PSD definition proposed for ADDS is identical to the PSD definition used in the HD Radio system [4]. Some key requirements for Program Service Data are:

- Program audio and associated data shall be transmitted synchronously, so receivers can acquire correlated audio and data at the same time.
- PSD messages shall not exceed 1024 bytes, including HD RLS overhead. 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 Reference [4]. 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.
- 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 up to the maximum ID3 tag size of 1,018 characters.

The PSD Transport provides a generic and reliable packet transport for implementing data services. This section describes the PSD Transport and PSD PDU generation.

5.2.3.1. PDU Format

A PSD PDU is contained in an HDLC-like frame delimited by Flags as shown in Table 14.

Table 14: 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 5.2.3.2
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 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.

5.2.3.2. Default PSD Packet Definition

Program Service Data uses the default packet format shown in Table 15.

Table 15: 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[ ]	1-1024	Payload length is variable up to 1024 bytes.

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.

The details of the PSD transport protocol and message definitions are provided in Reference [4]. The details of the message transport will not be repeated for this report.

5.2.3.3. PSD PDU – Example

Figure 5-6 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.

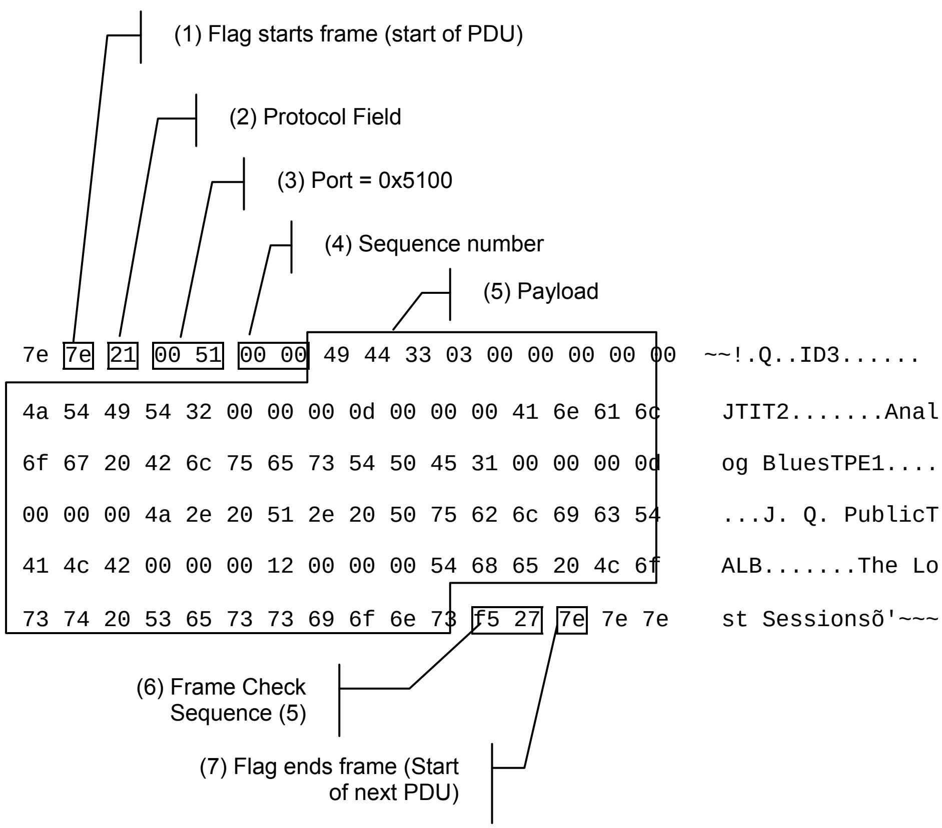


Figure 5-6: PSD PDU Example

6. Data Specifications

6.1. ID3 V2.3 for PSD

The AM Digital Data Service allows PSD Data to be transmitted along with the program audio. PSD Data is intended to describe or complement the audio program heard by the radio listener. The PSD definition proposed for ADDS is identical to the PSD definition used in the HD Radio system [4].

The following subsections provide:

- An introduction to the basic PSD Data content
- A description of broadcast PSD Data processing
- The format of PSD Data messages

6.1.1. PSD Data Content

PSD Data 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 Data fields include the following:

- Title
- Artist
- Album
- Genre
- Comment
- Commercial
- Reference Identifiers

A detailed description of the PSD Data structure is discussed in Subsection 6.1.3.

6.1.2. PSD Data Transmit Processing

PSD Data 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 Data is consistent. PSD Data providers input the desired content (e.g., artist, title, etc.) and transfer the resulting PSD Data message to the service interface.

Some key considerations for PSD Data are:

- Program audio and associated data must be transmitted synchronously, so receivers can acquire correlated audio and data at the same time.
- PSD Data messages are continuously transmitted with the most recent message transmitted repeatedly.
- PSD Data providers send a new PSD Data message when the PSD Data content has changed.
- PSD Data messages cannot exceed 1024 bytes.

6.1.3. PSD Data Message Format

PSD Data is formatted using a subset of the standard called ID3v2 (see reference [5]). 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 AM Digital Data Service uses ID3 to deliver program associated data along with real-time broadcast audio.

PSD implements a specific subset of the ID3v2 parameters. The ID3v2 general structure is as follows:

- The 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 (e.g., 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 6-1 shows the general ID3 message structure.

PSD Data utilizes only the identified subset of the defined ID3v2 standard. Broadcasters providing PSD Data must, at a minimum, transmit the Title and Artist information. Table 16 gives a description of ID3 frames used for PSD Data. For a detailed description of ID3v2 message encoding used for PSD Data, see Reference [5].

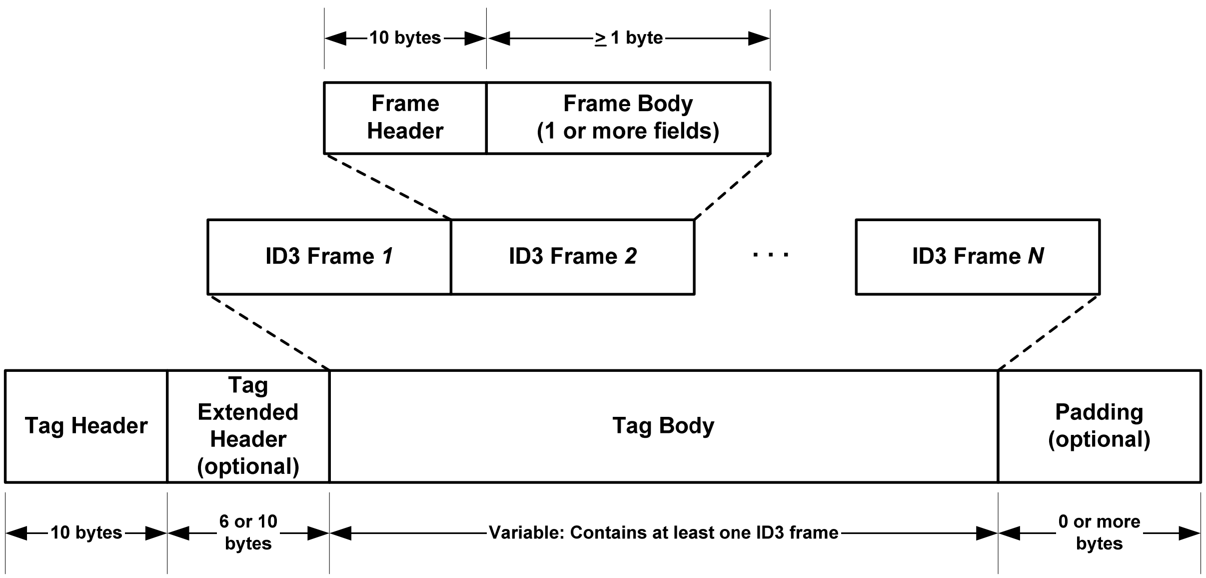


Figure 6-1: ID3 Message Structure

Table 16: ID3 Frames Supported by PSD Data

	PSD Data	ID3	ID3	Description		Type	
	Attribute	Frame ID	Field		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.	e.g., (8) Jazz, (17) Rock (32) Classical	(101) Speech	(101) Speech
5	Comment	COMM	Short description field	One-line Title for Comment Description	Comment Title	Comment Title	Comment Title

	PSD Data Attribute	ID3 Frame ID	ID3 Field	Description	Type Music	Type Talk	Type Announcement
5	Comment	COMM	Content field	Comment Description. Detailed explanation, user callback information or further information.	web site, contact or other info	Talk Show call-in number, or other show info	Announcement or Advertising statement, Point-of-sale or more info
6	Commercial	COMR	Price	Price of merchandise	The commercial frame facilitates sale of products and services. (Note, the binary picture is not required.)		
			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)			
			Name of seller	Text identifying seller			
			Description	Textual description of advertisement			
			Picture	Picture of advertised item			
			Seller logo	Binary graphic of seller logo			
7	Reference Identifier	UFID	Owner Identifier	This ID is used to: 1. Broadcast a single PSD Data message as one or more separate PSD Data messages. 2. Provide additional PSD Data identification and cross-referencing for future enhancements.	If Owner Identifier is set to PADLINK, then the identifier contains a unique message identifier, which allows a single PSD Data message to be broken into one or more PSD Data messages for broadcast transmission. Each individual PSD Data message is a complete decodable message, but may only contain a subset of the PSD Data content. For example, the title, artist and album may be transmitted in one PSD Data message and the commercial information may be transmitted separately. Both messages contain a common unique identifier, which is the mechanism for correlating the complete PSD Data message. Collectively the two messages represent a unified PSD Data message. The combined messages should be treated as a whole, when performing receiver-side processing. The identifier field contains a unique number ranging from 0 to 65,535. This number is unique only to the Main Program Service instance of a given station.		

7. Industry and Product Impacts

7.1. Transmission

The AM Digital Data Services is intended for analog AM stations that have not converted to HD Radio broadcasting. As such, the requirements for a radio station transmitting this limited digital service will be less than for a full HD Radio broadcast.

A generalized broadcast configuration is provided in Figure 7-1 below. This architecture defines the basic functional blocks required. The GPS Timing Reference and AM Exciter will have added requirements to support synchronization and modulation of the AM Digital Data Service. Complete specification and detailed design for these components will require further investigation and study.

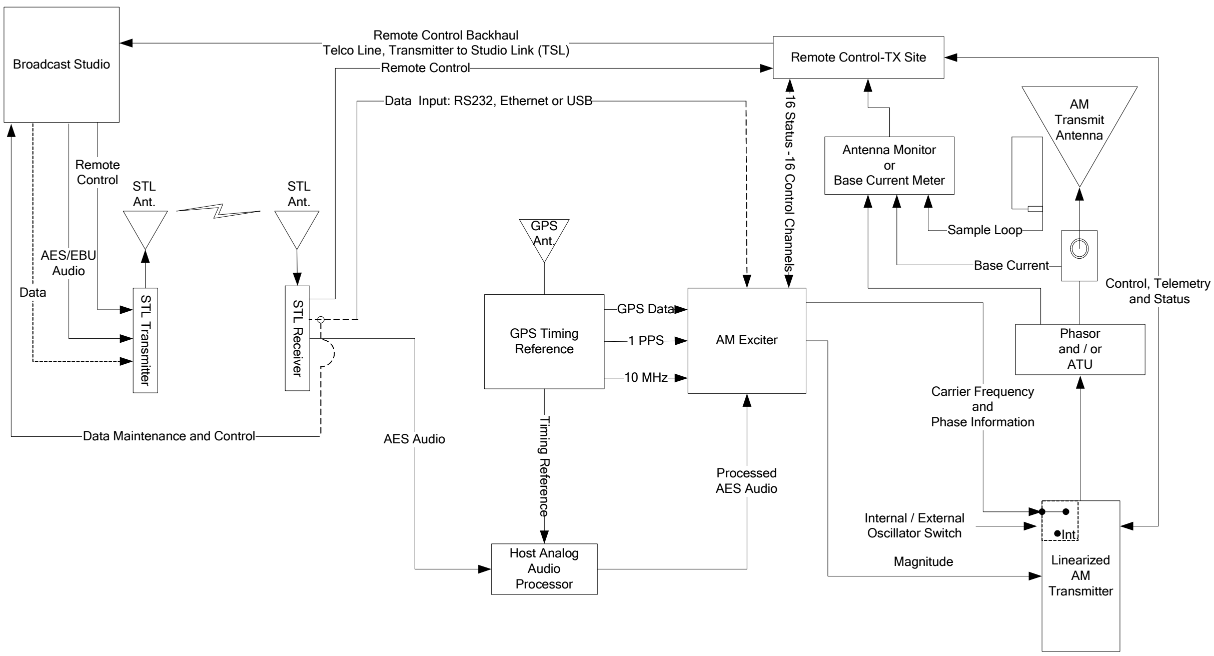


Figure 7-1: Generalized AM Broadcast Configuration

7.1.1. Broadcast Studios Interface

To support AM Digital Data Services, an AM transmission system will require capability to support PSD text generation and real-time station information services. It is recommended that the broadcast studio be equipped with studio automation equipment which supports HDP PSD SDK V4.7 or later. (Note: HDP PSD SDK V4.7 refers to the HD Radio Protocol for PSD transport into the broadcast equipment. This software application protocol definition is proposed for the ADDS implementation).

The broadcast studio should have a data services line to the transmitter to convey the content. A phone link with 9600 BAUD modem to the transmitter site would be sufficient to support the data required at 300bps. If available, STL service would provide more robust capacity.

7.1.2. GPS Timing Reference

The AM Digital Data Service system will accept a feed (optional) from a Global Positioning System (GPS) clock source. The GPS signal reduces AM Data digital channel receiver lock time by introducing a highly accurate time reference. Since the GPS signal guarantees specific frame timing, the system can perform relatively minor tracking synchronization and avoid going through a complete synchronization cycle.

Additionally the use of a known standard time reference could simplify channel to channel cooperative data exchange capabilities and many other data service features.

The GPS receiver provides 1-PPS and 10-MHz Timing to the AM Exciter. These signals provide internal synchronization for the exciter and phase-lock reference. Time of day and other control information is also provided via an RS-232 link.

7.1.3. AM Exciter

To implement the AM Digital Data Service, a simplified version of an HD Radio Exciter is required. The system is designed to support a low-cost implementation at a radio station. Processing text messages and modulating three data subcarriers greatly reduces the transmission complexity as compared with a full IBOC AM broadcast system.

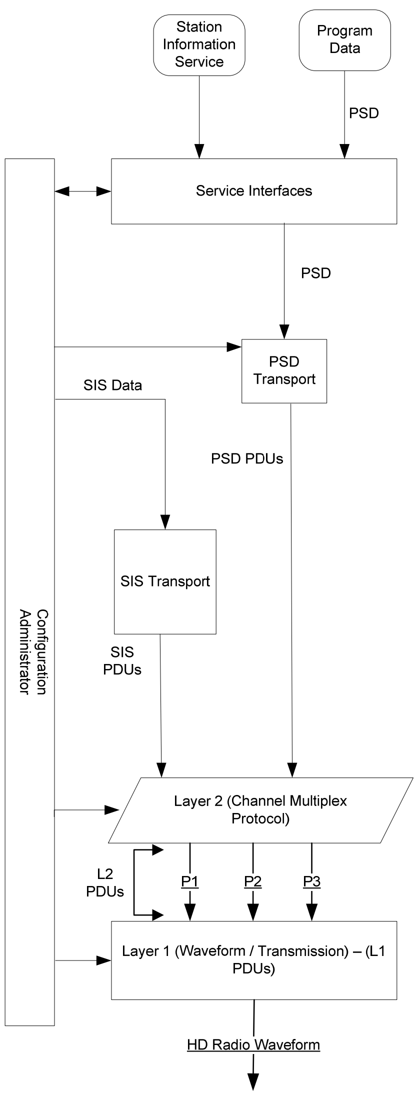


Figure 7-2: AM Digital Data Service Protocol Layers

7.1.3.1. Exciter Interfaces

The ADDS Exciter shall have a data interface to support text message input in support of station alerts or PSD transmission. The data interface may be RS-232 serial interface for reduced complexity. Alternatively, an Ethernet connection may be implemented. However an Ethernet interface would increase infrastructure needs at the transmitter site.

The ADDS Exciter shall have time references for signal synchronization. An interface for the 10-MHz reference clock and the 1-PPS reference time sync are required to manage waveform timing and framing alignments. All reference signals shall be locked to a GPS time reference.

7.1.3.2. Exciter User Interface

The ADDS Exciter shall support a user interface (UI) for limited control or configuration. The UI shall provide capability for diagnostics, subcarrier power level adjustment, and subcarrier selection. Additionally, a configuration for station maintenance and MER signal evaluation shall also be provided.

7.1.3.3. Exciter Modulation Design

The Exciter modulator system shall implement the modulation techniques defined in the preceding sections. Because ADDS modulates three pairs of subcarriers, a IDFT can be utilized in place of the IFFT required for IBOC system implementation.

7.1.4. Linearized AM Transmitter

The ADDS system is designed to work within existing analog AM stations performance envelope and does not introduce more stringent broadcast criteria then exists already in the Code of Federal Regulation (CFR). A station meeting the performance criteria set forth in the CFR will meet the requirements of ADDS.

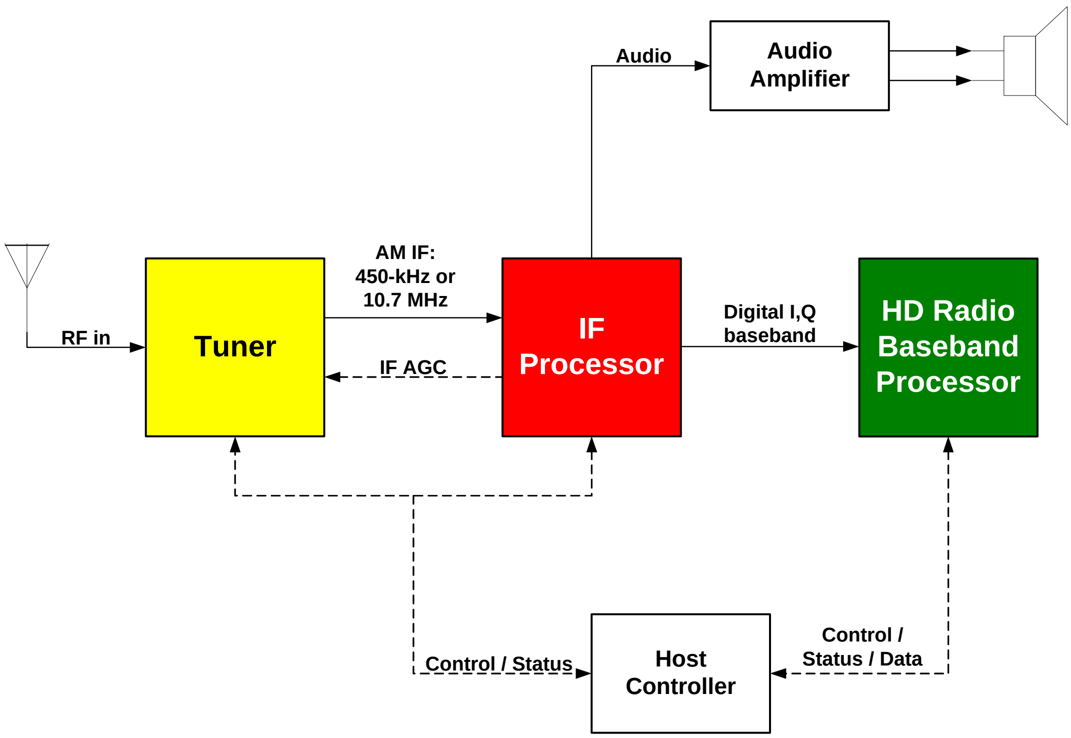
AM broadcast transmitters must provide minimal phase distortion to pass the BPSK/QPSK carriers. Group delay is important since the center carrier serves as a phase reference signal.

The transmitter must be capable of being driven by an external Exciter and timing source.

7.2. Receiver

The proposed AM Digital Data Service contains several modulation and transport components from the HD Radio broadcast specification. This design approach allows for a receiver architecture with many functional blocks shared with an HD Radio receiver. However, the AM Digital Data Service would have significantly simplified implementation with reduced memory and reduced computational requirements compared with a standard HD Radio receiver. It is envisioned that ADDS receiver could be added for no cost to future HD Radio receiver products

The receiver architecture for the AM Digital Data Service will consist of RF tuner, IF Processor, and IBOC Decoder (HD Radio Baseband Processor).



7.2.1. RF Tuner

The tuner system shall have at a minimum:

- an antenna input port
- a port for tuning control
- an output port

Additionally, an optional port for AGC control and noise blanking control may be included.

Where applicable, a carrier frequency of one MHz is assumed. IBOC reception of all modes is assumed. These requirements (Table 17: RF Tuner Requirements) are adopted from the HD Radio RF tuner requirements.

Table 17: RF Tuner Requirements

Parameter	Conditions	Requirement
Minimum Operating Signal	66 dBuV (-41 dBm) interference	25 dBuV (-82dBm)
Maximum Operating Signal	No interference	80 dBuV (-27dBm)
	40 dBuV (-67 dBm) first adjacent	
	60 dBuV (-47 dBm) second adjacent	
	80 dBuV (-27 dBm) third or greater	
Output Level	50 ohm load	+4dBm (RMS) > Level > -10dBm (RMS)
Equivalent Input Noise		1.25 nV/root Hz
Antenna referred IP2	Input level 66 dBuV (-41dBm)	> 138 dBuV (+31dBm)
Antenna referred IP3	Input level 66 dBuV (-41dBm)	> 102 dBuV (-5 dBm)
Frequency Accuracy	Quiescent	686 Hz
Settling Time		50 mS
Phase Noise		
	1 Hz offset	-33 dBc/Hz
	10 Hz offset	-39 dBc/Hz
	100 Hz offset	-80 dBc/Hz
	1 kHz offset	-90 dBc/Hz
	10 kHz offset	-100 dBc/Hz
	100 kHz offset	-110 dBc/Hz
	> 100 kHz	-110 dBc/Hz
Differential Group Delay	Variation over any 1-kHz band	<5 uS
AGC	Dynamic Range	>60 dB
	Attack time constant	50 mS
	Decay time constant	200 mS
Discrete Spurious	5-kHz bandwidth	< 0 dBuV (-110 dBm)

Optional noise blanking shall consist of:

- detecting noise pulses at RF or IF which are at a noise blanking threshold above the average signal strength in the vicinity of the pulse.
- excising noise pulses by setting the RF or IF waveform to zero for the duration of the noise pulse.

Noise blanking control may be provided to allow disabling of noise blanking. The following parameters shall apply to noise blanking:

1. Point of blanking — wideband (> 30 kHz) RF or IF prior to demodulation
2. Noise blanking threshold — 20 dB above pulse average value with an interval of 6 ms centered about the pulse
3. Noise blanking duration — reciprocal of the system bandwidth at the point of noise blanking (e.g., <20 μs for blanking at a 50-kHz wide IF point)

7.2.2. IF Processor

7.2.2.1. Performance Specifications

The baseband signal into the ADDS decoder shall have the characteristics listed in Table 18. These requirements combine analog-to-digital sampling and digital down-conversion. This functionality may be met in the RF tuner or other chipset implementing the IF Processor sub-block. A sample description for digital down-conversion is provided in Section 7.2.2.2.

Table 18: AM Digital Data Service Decoder Input Signal Requirements

Parameter	Specification
Nominal Frequency	10.7 MHz
Tuning error	+/- 0.1 parts per million
Spurious output	<-80 dBc where 0dBc refers to nominal carrier level
Insertion gain	0 dB
Output bit width	16 bits two's complement (S1.15)
Passband	0 to 10kHz
Passband ripple	<1 dB peak to peak
Stopband attenuation	<1dB @ 10kHz; >80 dB @ 46.5 kHz

Zero-dB insertion gain has the meaning that a full scale passband sinusoid at the input will result in a full scale sinusoid at the output. The input bit width shall accommodate the full bit width of the A/D output in any event, and the above specification shall not be construed so as to suggest truncating bits of an A/D converter which provides more bits than the above specified number.

7.2.2.2. Functional Description

The AM direct digital down converter (DDC) subsystem performs the following functions:

- accept high rate samples of the IF signal from the A/D converter.
- digitally unidirectional heterodyne the (real) sampled signal to (complex) baseband
- perform decimation filtering
- decimate the filtered complex baseband signal by an integer factor to a final sample rate sufficient to represent a single channel bandwidth of 20 kHz. The decimation factor is dependent on configuration. The desired output sample rate is 46.51171875 kHz.

Nominal insertion gain is defined relative to full scale signal values of pass band signals. The nominal insertion gain of the DDC as a whole shall be zero dB. This has the meaning that a full scale real pass band sinusoidal signal from the A/D shall result in a full scale complex exponential sinusoid at the output of the DDC.

The DDC shall contain, at a minimum, the following functional elements.

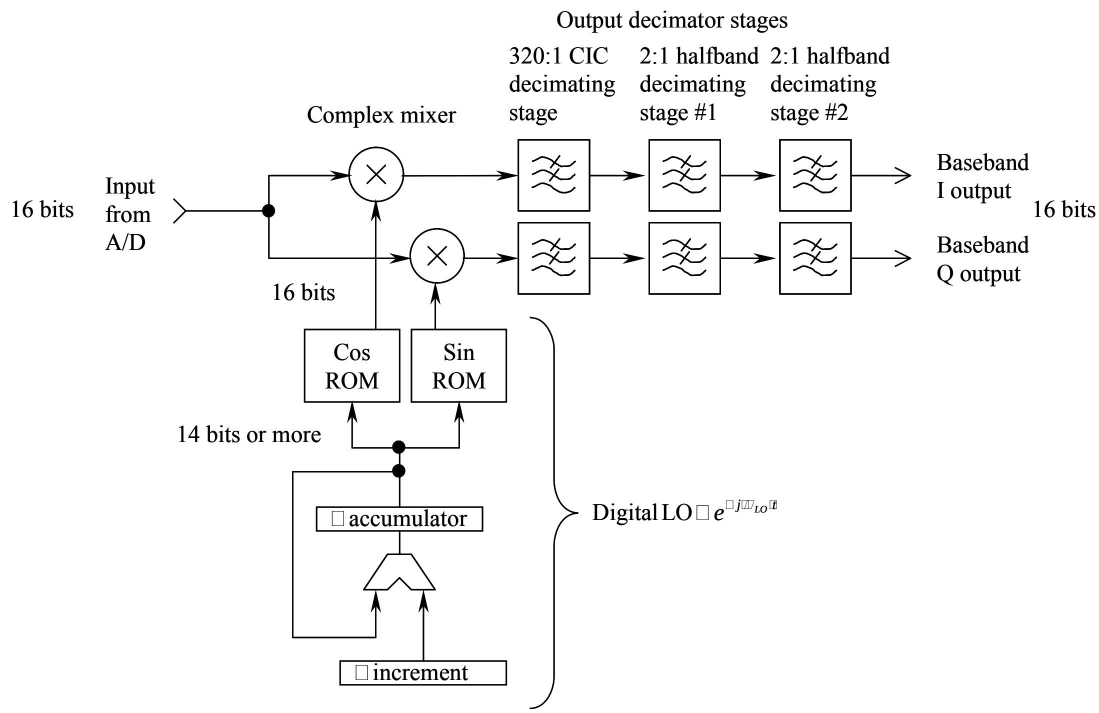


Figure 7-3: AM DDC block diagram

The digital local oscillator shall be a complex sinusoid of the form

$$I_{LO}(t) = \cos(\omega_{LO} \cdot t)$$
$$Q_{LO}(t) = -\sin(\omega_{LO} \cdot t)$$

where ω_{LO} is the IF frequency at which the signal of interest is centered, nominally 10.7 MHz. The LO frequency shall be rationally derived from, and synchronous with, the working clock of the DDC, nominally the A/D sample clock.

The digital complex mixer shall functionally consist of two digital multipliers which multiply the input signal by the digital LO signals $I_{LO}(t)$ and $Q_{LO}(t)$ respectively, producing a complex output signal which has the desired signal centered at baseband.

The decimation filters shall be real coefficient linear phase digital filters which have a passband centered at zero Hz, and multiple stopbands centered at integer multiples of the output (decimated) sample rate. Identical filters shall be supplied for the real (I) part and the imaginary (Q) part, respectively. The purpose of these filters shall be to condition the complex baseband mixer output so that destructive aliasing of the desired signals cannot occur when their outputs are decimated by the decimator which follows. The nominal insertion gain for signals within the passband of the decimation filters shall be 2 (6.01 dB). This is because the combined effect of the digital complex mixing operation and decimation filtering is to eliminate the hermitian conjugate of the (real) input signal from the A/D, thereby reducing the signal amplitude by two. The net insertion gain of the DDC as a whole will then be the required zero dB.

7.2.3. AM Digital Data Service Decoder

A simplified diagram of the ADDS decoder is provided in Figure 7-4. This proposed receiver architecture utilizes building blocks currently available in HD Radio decoder ICs. These same blocks could be added to other lower-cost decoder solutions with minimized processing.

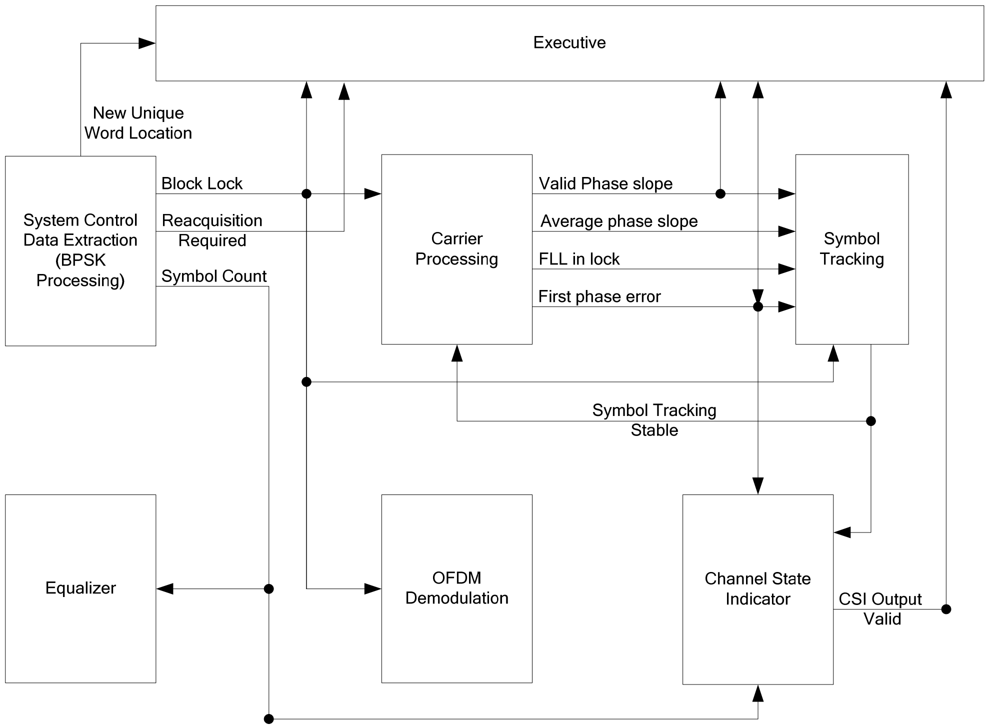


Figure 7-4: AM Digital Data Service Decoder Architecture

7.2.3.1. Decoder Functional Description

The ADDS functional decoder shall implement algorithms and processing to decode the IBOC subcarriers. Basic functionality in the receiver will include:

- Subcarrier acquisition
- BPSK synchronization and processing
- Subcarrier tracking
- Symbol tracking
- OFDM demodulation (simple DFT)
- SIS or SMS transport processing
- PSD transport processing

If 16-QAM modulation is implemented, then some level of equalization and channel state detection will be required.

The ADDS receiver will require a baseband interface for sampled I/Q data samples from the digital down-converter. The interface shall support a sample rate of 46511.71875 samples per second. However, other input sample rates can be supported if a sample rate converter is present in the decoding IC.

7.2.3.2. Processing Requirements

The ADDS receiver can utilize many functional blocks from the HD Radio decoder. Current estimates of processing requirements (based on HD Radio functionality) indicate a moderate level of processing is needed for the ADDS receiver.

The processor shall decode OFDM symbols at a rate of 172.3 symbols per second. Each symbol consists of 270 samples. The processing requirements are dominated by the carrier tracking and OFDM demodulation.

- Processing estimate: 29 Million Instructions per Second (MIPs)
- RAM memory estimate: 24 kBytes
- ROM memory estimate: 2.5 kBytes

In most applications, ADDS processing will be implemented in HD Radio decoder ICs. Future options may include a low cost, stand-alone decoder function implemented in a general purpose processor or inexpensive DSP.

8. Summary

This report summarized proposed use cases and modulation techniques for implementing a low-cost digital data service on analog AM stations. The proposed system utilizes several building blocks from the IBOC standard defined by the NRSC for AM digital broadcasting. The proposed system provides a capability for transmitting low-rate text services. It is not designed to be a full digital broadcast and does not provide capability for transmission of digital audio or other advanced services. This system is intended to enhance analog broadcasting with limited capability and provide a transition to a full digital broadcast implementation.

A general modulation scheme is proposed based on the IBOC system standard. The study analyzes different OFDM modulation techniques and reviews reception/performance impacts of proposed options. A maximum of three OFDM subcarrier pairs are defined to provide data transport capability of up to a maximum of 1098 bits per second. The proposed OFDM subcarriers would be transmitted under the analog audio broadcast at power levels to minimize any self-interference.

Several use cases and transmission options are suggested. Each set of options has an impact on the final system design parameters. A number of performance characteristics need to be considered before final system definition. The AM broadcast groups should consider:

1. Data use cases
2. Subcarrier power levels and their impact on coverage and self interference
3. Requirements for data throughput
4. Receiver acquisition times and fast tune/scan capability

Addressing these key areas will ultimately achieve industry standard requirements for coverage and transport capability.

A number of implementation requirements have been defined in this report. The study reviewed possible impacts to the broadcast infrastructure at a radio station and to the receiver architecture in a receiver. Additional details for these major implementations require further study and definition. Any future work should include:

1. Field test cases and trial implementations
2. Transmitter/Exciter design
3. Receiver algorithms and implementation

The technical details of this report were provided by Dr. Brian Kroeger, Marek Milbar, Jeff Detweiler, Russ Mundschenk, Mike Raide, Tom Walker, and Ashruf El-Dinary. Technical documentation support was provided by Stephen Dowd.

iBiquity Digital thanks NAB FASTROAD for the funding to investigate potential solutions for an AM data service.