Before the
Federal Communications Commission
Washington, D.C. 20554

In the Matter of:	)	
	)	
Digital Audio Broadcasting Systems	)	MM Docket No. 99-325
And Their Impact on the Terrestrial	)	
Radio Broadcast Service	)	

Reply Comments of David L. Hershberger on National Radio Systems Committee's "In-Band/On-Channel Digital Radio Broadcasting Standard NRSC-5"

The comments of Microsoft Corporation, Broadcast Signal Lab, LLP, and Impulse Radio correctly point out that NRSC-5 represents an archaic form of standardization. I wish to heartily endorse the views of Microsoft et. al., but I also wish to point out in these reply comments that the notion suggested by Microsoft should be expanded.

Microsoft suggests that the audio codec used for digital audio broadcasting should be upgradeable. I would add to this that the modulation method should also be upgradeable. In other words, digital radio broadcasting should be done with a purely software defined radio (SDR) approach, and not the restrictive, secretive, and monopolistic proposal at hand.

NRSC-5 proposes to standardize the transmitted signal. This is archaic, vacuum-tube era thinking. Today, general-purpose digital signal processors are used to decode digital radio signals, and in some cases, analog signals. These processors are reprogrammable. It would be much better to instead *standardize the receiver*. The receiver should be characterized in terms of its instruction set, RAM, NVRAM, clock speed, etc., and a secure over-the-air programming method. Then, broadcasters may choose a modulation method and audio codec, which best suits their needs. This approach would not become prematurely obsolete the way NRSC-5 most certainly will.

NRSC-5 is archaic because it proposes to standardize the transmitted signal. For all time, the standard will be limited to just one audio encoder and just one modulation method (although there may be some minor variations in each). This has already proven to be a shortsighted concept in Europe where Eureka 147 DAB has been deployed. The audio codec (MPEG-2) is already obsolete, by several generations. If the audio codec were upgradeable, multicasting could be significantly increased.

At the 1997 IEEE Broadcast Technology Symposium, I presented a paper detailing how a SDR approach to digital radio broadcasting might be implemented. That paper is included with these reply comments. (Please note that I am no longer employed by

Continental Electronics, as I was when the paper was presented.) There are many ways to implement SDR, and the purpose of the paper is to give an example.

This paper was written in 1997 just as DRM committee work was beginning. Now that DRM is being deployed, I would suggest that one of the most robust DRM transmission modes could be used as the method for transmitting demodulation software to SDR receivers, rather than the "straw man" in the 1997 paper.

Most of these reply comments concern the AM system but they are also generally applicable to the FM system.

NRSC-5 is fatally flawed for the following reasons:

Excessive bandwidth and interference

NRSC-5 occupies three channels rather than one, causing adjacent channel interference on a wide scale, particularly in the AM system. The AM signal format is incompatible with the allocation structure, especially at night. To support the comments I made in this proceeding in June 2004, I had posted short MP3 files (slightly less than three minutes) showing what the interference sounded like during the March 2004 nighttime IBOC tests at KXNT in Las Vegas on 840 kHz, which interferes with reception of our local 5 kilowatt AM station on 830 kHz (KCNO, Grass Valley, CA). Those recordings are still available online. This file was recorded on March 3, 2004, shortly after 10 PM local time:

http://www.w9gr.com/kncoiboc.mp3

This file was recorded several days earlier (February 23, 2004) when KXNT was not testing IBOC, also shortly after 10 PM local time:

http://www.w9gr.com/knco.mp3

These recordings were made 6.4 miles from the KNCO transmitter and 384 miles from the KXNT transmitter. For more details on the receiver, directional antenna patterns, etc., please refer to my June 2004 comments.

I believe that KXNT's STA for nighttime IBOC testing expired June 20, 2004. Yet I have heard nighttime IBOC transmissions from KXNT on many occasions between sunset and about midnight local time when I have tried to listen to KNCO, apparently because KXNT sometimes forgets to turn off IBOC at night. Here is a partial list of dates on which I have heard nighttime IBOC interference from KXNT:

July 18, 2004	August 21, 2004	June 21, 2005
July 19, 2004	August 22, 2004	June 26, 2005
July 20, 2004	September 20, 2004	June 29, 2005
July 24, 2004	September 21, 2004	July 6, 2005
July 29, 2004	January 23, 2005	July 15, 2005
August 4, 2004	March 5, 2005	July 16, 2005
August 11, 2004	March 6, 2005	August 6, 2005
August 12, 2004	March 20, 2005	
August 15, 2004	June 4, 2005	

In these cases I have heard strong noise-type interference affecting only KNCO's lower sideband, and affecting only KOA's upper sideband on 850 kHz. Therefore, the interference is originating from an IBOC transmitter whose analog signal is on 840 kHz, and which is transmitting IBOC interference sidebands from 825-830 and 850-855 kHz.

This is just one example of wide scale interference to local reception that will occur if NRSC-5 is approved for nighttime operation on AM. I would expect the interference to be considerably worse in the eastern half of the country where stations are packed more closely together.

The analog and digital signal arrangement is inappropriate

The NRSC-5 proposal places the digital signals where they create the maximum possible interference to adjacent channel analog signals. The analog signal is on the station's assigned frequency, while the digital sidebands occupy both adjacent channels, creating demodulated interference in a 0-5 kHz bandwidth.

There are at least three possible solutions to this problem:

1. Move the digital sidebands away from the carrier frequencies of adjacent channels. The digital sidebands as proposed occupy ± 10 to ± 15 kHz. For example, if the digital sidebands occupied ± 2.5 to ± 7.5 kHz, then the interference generated would be above 2.5 kHz and out of the bandpass of AM radios claimed by the proponents of NRSC-5. In wider bandwidth radios, the digital sidebands would interfere with the originating station rather than "innocent bystander" adjacent channel licensees.

2. Swap the analog and digital signals. For example, for an AM station assigned to 840 kHz, the digital signal would be centered on 840 kHz, occupying 835-845 kHz. Two analog signals would be transmitted at lower power on 830 and 850 kHz. For spectral control, and to minimize interference to second adjacent channels, these analog signals should be transmitted as linear vestigial sideband (VSB), with an outer vestigial sideband of approximately 500 Hz and the full sidebands folding inward. Since most of the energy in AM signals is at low frequencies, this would minimize envelope detector distortion while confining the total occupied bandwidth to just 21 kHz rather than the 30 kHz

occupied by NRSC-5. Analog listeners would tune to one of the analog signals on the adjacent channels.

(Having analog listeners retune is not without precedent. WNZK, 680/690 kHz, is a dual-frequency AM in Detroit. WNZK listeners must retune their radios between day and night. WNZK operates on 690 kHz during the daytime and 680 kHz at night.)

This approach would have several benefits. First, the interference to adjacent channels would be in the form of analog signals rather than the caustic, raucous digital signal. Second, interference to second adjacent channels would be eliminated. Third, for directional stations, the antenna pattern and its protections would be maintained for the digital signal. Fourth, the power allocated to the digital signal could be made much higher, allowing much better digital performance. Fifth, in cases of severe adjacent channel interference, one of the two analog signals could be deleted.

3. Also allow analog-only broadcasters to occupy three channels instead of one, as a means of spectral self-defense. If digital broadcasters get three contiguous channels, then it is only fair that analog broadcasters have the same amount of spectrum. For example, if an analog-only broadcaster on 830 kHz experiences IBOC interference, he could transmit additional analog signals on 820 and 840 kHz at a lower power level than his 830 kHz signal. Listeners could tune to whichever of the three frequencies suffers the least amount of interference. The outer signals on 820 and 840 kHz would again be vestigial sideband AM, however in the case of analog broadcasts, they would be folded outwards rather than inwards. In this example the 840 kHz signal would be primarily upper sideband and the 820 kHz signal would be primarily lower sideband, limited to 5 kHz of audio response with a 500 Hz vestigial sideband. The 830 kHz signal could still be transmitted with up to 10 kHz of bandwidth. The result would have the same bandwidth as NRSC-5 (30 kHz), but the nature of the adjacent channel interference generated by the two additional analog signals would be less objectionable since they would be analog.

This last suggestion may seem sarcastic and outrageous, but that is only because the whole notion of NRSC-5 is also outrageous in many ways.

The audio codec cannot be upgraded

NRSC-5 would forever limit both the AM and FM systems to a single audio codec. Audio coding technology is improving rapidly and it would be negligent to allow only one audio codec to be used, even after it is obsolete.

The modulation system cannot be upgraded

Although advances in modulation systems are not proceeding as rapidly as audio coding developments, there is still progress being made, which would be denied if only one basic modulation method is allowed.

The digital modulation system proposed is not sufficiently hierarchical

A digital modulation system should degrade gracefully in the presence of reduced signal to noise ratio, interference, etc. This is particularly important for the AM system. Yet NRSC-5 degrades abruptly, at which point it must switch to the analog signal. This characteristic illustrates that the system still needs work. The digital modulation and coding systems should include both robust and non-robust components, with a degraded minimal signal carried by the robust modulation components. It should never be necessary to revert to the analog signal, especially since the analog signal is expected to eventually disappear.

The audio codec and modulation system are secret and proprietary

Creating an FCC broadcast standard that is secret and proprietary would be unprecedented. The technical descriptions of the FCC-standardized pilot tone FM stereo system, the BTSC system, the NTSC system, and the ATSC system are all public. After the patents expire for each of these systems, they become public domain. This is not the case with the secret NRSC-5 system. Unless these systems are completely disclosed, users will be forced to pay royalties beyond the patent terms for the privilege of using a trade secret. This would be a perpetual windfall for those doing the licensing, and an ongoing burden for the public.

NRSC-5 describes the transmitted signal rather than the receiver

We are now well past the time when low-cost software-defined radios (SDRs) can be implemented for broadcast reception. The public interest demands that digital audio broadcasting be done this way. Automatic software upgrades to receivers can be performed over the air, invisibly to the consumer. This will allow improvements in audio quality, addition of multiple program streams, extension of coverage area, reduction of interference, tailoring the signal and codec characteristics to each broadcaster's needs, and additional services that cannot be foreseen at this time.

Just as open-source audio codecs have been developed under the General Public License (GPL), public domain digital broadcasting standards may also be developed which will be royalty-free for both broadcasters and receiver manufacturers. This would certainly serve the public interest better than a secret monopoly with avaricious royalties.

In the early 1980s the Commission tried a "marketplace" approach to AM stereo, allowing any kind of AM stereo signal to be transmitted. Whether this worked is open to debate, but most would admit that it failed. In any event, it was done at a time when consumer electronics technology did not support the concept of economical and accurate multisystem receivers, let alone reprogrammable receivers. That is no longer true. Reprogrammable digital signal processors have been used in broadcast radios for several years now, so the notion of a software-defined AM/FM radio has been quite feasible for years.

Conclusion and recommendation

I recommend that the Commission send the NRSC-5 proposal back to committee with the stipulation that NRSC rework the system to be fully software defined in such a way that all receivers may receive any kind of transmitted signal. This will bring competition and further much-needed improvements to digital audio broadcasting. The alternative is high costs, extensive interference, and technical stagnation.

If approved, NRSC-5 will be obsolete soon after its approval.

Sending NRSC-5 back to committee for improvements will also satisfy Microsoft's useful suggestion that the audio codec be made reprogrammable. At the same time, NRSC can make the modulation method(s) reprogrammable too.

With these improvements, if broadcasters wish to transmit the NRSC-5 signal as secretly specified today, they may do so. But if they wish to use an improved signal format, they will have that option also.

Respectfully submitted,

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Attachment: 1997 IEEE BTS paper

A Proposal for a Software Based LW/MW/SW Digital Broadcast System

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ABSTRACT

Proposals are currently being made for digital broadcasting on the AM bands (longwave, medium wave, and shortwave). Although it would be desirable to develop a modulation and compression system which would be optimum for all users for many decades, it is unlikely that an unchanging optimum system will ever exist for all purposes and all propagation conditions.

Unlike previous standardization efforts where the broadcast signal itself was specified, it is now possible to instead standardize the receiver. The receiver would include a low cost general purpose digital signal processor (DSP). It could be programmed using a low speed ancillary data channel included in off-air signals to decode any of several different digital modulation coding and audio compression systems. Receivers would store new decoding software in nonvolatile memory and would execute it when instructed to do so by the broadcast station. These functions would be transparent to the receiver's user. New modulation and compression methods could be implemented as they are developed, avoiding premature obsolescence.

A software based receiver would allow the selection of the optimum transmission method for each station. The transmission method could change with time of day to compensate for interference and/or skywave propagation, and would allow trading off coverage area, audio quality, skywave performance, time delay, graceful degradation, etc.

The remainder of this paper proposes possible architectures for the receiver, and a packetized signaling system to control and update the receiver.

INTRODUCTION

A world consortium for digital AM radio broadcasting has been formed, known as Digital Radio Mondiale or DRM. The purpose of the DRM consortium is to develop a world digital broadcasting standard for the AM bands below 30 MHz.

(Although a misnomer, digital broadcasting on longwave, medium wave, and shortwave - the traditionally AM bands - is being called "digital AM." "Digital AM" will be used in this paper to refer to digital broadcasting below 30 MHz.)

One of the requirements specified by the DRM consortium[1] reads as follows:

"Ideally the system should be designed in order that future improvements in audio compression algorithms can be implemented at the broadcasting side. Receivers in the field do not have to be replaced when using a new algorithm."

It is the purpose of this paper to present several techniques to make the digital AM system conform to this requirement. Basically, the receiver will be specified to include a general purpose digital signal processor (DSP). The required DSP decoding software will be broadcast along with the digital audio program.

This kind of system will provide a high degree of flexibility.

THE OLD METHOD:
STANDARDIZE THE SIGNAL

In the past, standardization activities have concentrated on the specification of a broadcast signal (Figure 1). Once the signal is defined and receivers are built for that particular signal, it is difficult to incorporate improvements. As technology improves, then the standard is either abandoned, or inflexibility limits its usefulness.

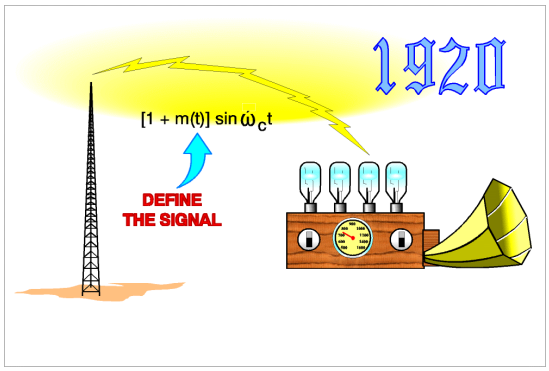

Figure 1 – Standardization by Defining the Transmitted Signal

Standardization activities often include highly competitive "systems battles," where considerable resources are expended in political as well as technical pursuits, promoting one's own technologies and attacking those belonging to others. The best technology does not necessarily emerge victorious. A software based system would eliminate "systems battles."

The old way of specifying the broadcast signal has been called "6SN7 thinking" (Figure 2). (For younger readers, "6SN7" refers to a type of vacuum tube developed well over half a century ago[2].) When the 6SN7 was in use, the only practical approach to standardization was to specify the transmitted signal.

Figure 2 – Archaic “6SN7 Thinking”

THE MODERN METHOD:
STANDARDIZE THE RECEIVER

Nowadays it is possible to instead standardize the receiver, allowing several different transmission methods to be used, invisibly to the receiver's user (Figure 3).

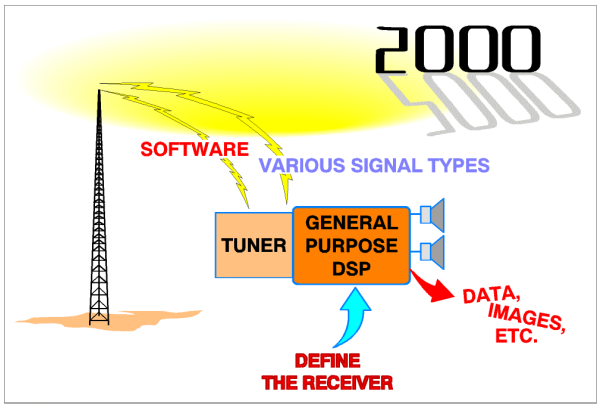

Figure 3 – Standardizing the Receiver Instead of the Signal

Regardless of what kind of digital AM broadcasting system is eventually implemented, the decoding will most likely be done with a general purpose DSP chip or core, as opposed to a dedicated single function hardware design. General purpose DSP chips are inexpensive, and allow fast development and easy product improvements.

An example of consumer products using general purpose DSPs is certain car radios, which use a Motorola DSP chip set to do IF filtering, AGC, and demodulation of analog AM and FM stereo broadcasts[3].

Another example is computer modems. Several manufacturers have produced 33.6 kbit/second modems which may be software-upgraded in the field to 56 kbits/second. The modem example is particularly pertinent because a large part of a digital AM radio will be the decoding portion of a modem.

An example of an existing system where the receiver is specified is MPEG. The MPEG video standards are essentially specifications for decoders, not for an exact method of signal encoding. The MPEG standards define the operation of the decoder, and it is up to encoder manufacturers to develop their own methods of generating a compressed signal. The MPEG standards do not tell an encoder manufacturer exactly how to produce motion vectors, for example. This kind of standard allows different manufacturers to compete in providing different kinds of tradeoffs: performance, artifacts, cost, etc.

Even if the radio is single-function (AM radio only), the DSP could be expected to be basically general-purpose in nature. If the AM radio is part of a consumer item with other functions, such as a television receiver, a Compact Disc player, or a digital audio player which uses some sort of compression, there would be even more reason to use a general-purpose DSP because it could perform these various functions by executing different software.

The standardized digital AM receiver would consist of a tuner followed by a general purpose digital signal processor (DSP), with a predefined method of transmitting decoding software to the receivers (Figure 4). Digital AM receiver operation would be controlled by software, downloaded to the receivers over the air. The modulation and compression types could be almost anything. As new modulation coding and audio compression algorithms are developed, they could be coded into the receiver's known machine language and broadcast along with the program material. Importantly, it would allow new techniques to be used almost immediately after they are developed.

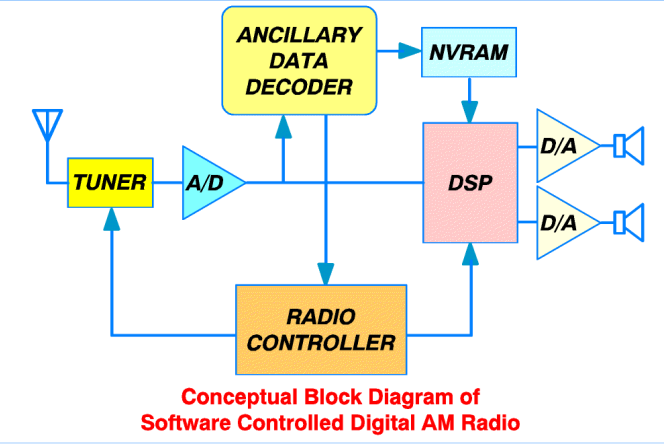

Figure 3 – SDR Broadcast Receiver

A software based digital AM system could actually advance the introduction of digital AM by several years. The tasks of receiver design and signal specification could proceed in parallel rather than serially. The digital AM process would not have to be slowed by a long comparative investigation of different modulation types and audio compression methods.

OBJECTIVES FOR
DIGITAL AM BROADCASTING

The objective of the DRM is to select a "...single, unique, tested, non-proprietary, evolutionary world-wide standard for digital broadcasting in the broadcasting bands below 30 MHz." To the extent that it is unlikely that a single modulation type and a single compression type will be clearly superior for all users on all frequencies under all propagation conditions for all time, a software based system becomes attractive.

The objectives of broadcasters and receiver users vary widely. Some broadcasters only care about their groundwave coverage area. HF broadcasters only care about skywave propagation. Some broadcasters have interference problems while others do not. Propagation conditions, interference, and antenna patterns frequently change with time of day.

A software based system would allow broadcasters to tailor the transmitted signal to best serve their listeners. There are many tradeoffs. Important ones include:

1. Daytime/nighttime modes. A station may transmit a higher data rate, with less forward error correction (FEC), and better audio quality due to lower compression during the daytime. At night, particularly if the station reduces power and/or uses a more restricted directional pattern, the tradeoff could be changed to make the signal more robust, but at the expense of a lower data rate, higher compression, and reduced audio quality.

2. Audio quality tradeoffs. Some of the proposed systems for digital AM broadcasting provide audio payload data rates of 6-48 kbits/sec[4], 32 kbits/sec[5], 20 kbits/sec[6], or up to 48 kbits/sec[7]. At these bit rates, the presently available audio compression algorithms provide audio which is perceptually lower than Compact Disc quality. The modulation method can be changed to provide a higher data rate and higher audio quality but at the expense of coverage area. Conversely, the coverage area can be improved but at the expense of data rate and therefore audio quality. Similarly, tradeoffs can be made between coding (and decoding) time delay versus audio quality.

3. Short fade immunity versus time delay. If a station serves a mobile audience in an area with many highway overpasses, or if an electrical storm is passing through the area, time interleaving can be increased to transform the burst errors from short fades or lightning into correctable errors. However, this tradeoff

would necessarily increase time delay. Unless redundant, non-delayed audio were transmitted, there would be a short delay from the time the station is selected until audio output from the receiver begins.

4. Adjacent channel interference. A station may broadcast a signal with more robust encoding (lower data rate) on the sideband with the most interference, and a higher data rate on the sideband which has less interference.

AUDIO COMPRESSION ALGORITHMS

Presently available audio compression algorithms require approximately 100 kilobits per second to provide audio quality comparable to FM stereo under good reception conditions, and somewhat more to sound similar to a Compact Disc. At the data rates which can be presently supported by digital AM systems, the audio quality will be somewhat worse. This has several implications.

First, unless compression technology improves significantly prior to the introduction of digital AM, at its introduction digital AM will not have audio quality comparable to FM or to Compact Discs.

Second, it will be highly desirable to have a digital AM system which will allow improvements in audio compression algorithms to be incorporated without changing the receivers.

Third, audio quality improvements through future compression technology advances will be possible only if receivers are easily field reprogrammable.

Audio compression technology seems to be progressing faster than modulation coding technology at this time. Even if a totally new audio compression technology (such as fractal audio) is developed, as long as it can be coded for the receivers, it can be implemented almost immediately.

ANCILLARY DATA SYSTEM

The signaling system used to transmit the decoding software to the receivers will be called the "ancillary data channel" in this paper. This would be a low speed signaling system which would occupy roughly 1-10% of the channel capacity. The other 90-99% of the channel would be occupied by the broadcast program material (which may include data, images, etc. in addition to audio programming).

When a receiver is tuned to a digital AM station, it first looks for the ancillary data channel. Ancillary data informs the receiver what the modulation type is, and what kind of audio compression the station is using. In most cases, the receiver will look in its nonvolatile memory (NVRAM) to find the particular decoding software for the station. If it finds it, it will then begin decoding the broadcast. In the rare situation when the receiver does not find the proper decoding software, it will download the decoding software to its NVRAM from the ancillary data channel. While the software is downloading, the receiver may provide an indication to the user that it is in a "learning" mode and that decoding will commence shortly. When the software has been downloaded, the receiver's audio outputs will unmute.

In most cases, however, use of new modulation formats will be preceded by software downloads beginning perhaps a month ahead of time . By that time, most receivers will have been updated, so that when a new modulation type is broadcast, the receivers can already decode it.

Broadcasters may also wish to cooperate by transmitting all publicly known decoding algorithms so that when tuning to a new station, the receiver will already know how to decode it.

Receivers which include other functions, such as digital audio broadcasting (DAB) receivers, computers, television receivers, etc., could receive software updates via alternative modes, such as FM broadcast subcarriers, the internet, or data included with TV or DAB.

Many receivers, including some battery operated types, may include a software search function. When the receiver is not in use it may periodically turn itself on and scan for stations using new transmission methods. If it finds a new transmission type, it would download the software for that kind of broadcast.

As transmission methods become obsolete, receivers could be made to sense the lack of use of a particular algorithm over time, and old decoding software could be automatically purged to make room in NVRAM for new software.

ANCILLARY DATA MODULATION METHODS

All of the candidate technologies for ancillary data transmission are well understood but some cooperation will be required among standardization bodies to define the method for transmitting decoding software to the receivers. This paper includes such a "straw man" proposal, intended as a starting point for discussion.

The ancillary data channel could be transmitted using a simple and robust bilevel phase shift keyed (BPSK) modulation system. The parameters of this signal would be adjusted depending on the program channel's basic modulation method.

At this time there are primarily two basic modulation methods for the program data: single carrier and multiple carrier modulation.

An example of a single carrier modulation system is quadrature amplitude modulation (QAM). Orthogonal Frequency Division Multiplexing (OFDM) is an example of a multiple carrier system.

The ancillary data channel would operate at the station's "carrier frequency," meaning the frequency where the carrier would be if the station were broadcasting a conventional analog signal. Two methods of multiplexing the BPSK ancillary data channel modulation are proposed - one for single carrier and one for multiple carrier systems.

For single carrier modulation, the ancillary data channel would be time division multiplexed (Figure 5). For multiple carrier signals, the ancillary data channel would be frequency domain multiplexed (Figure 6).

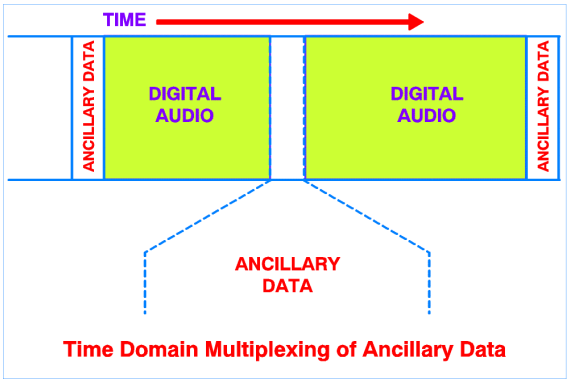

Figure 5

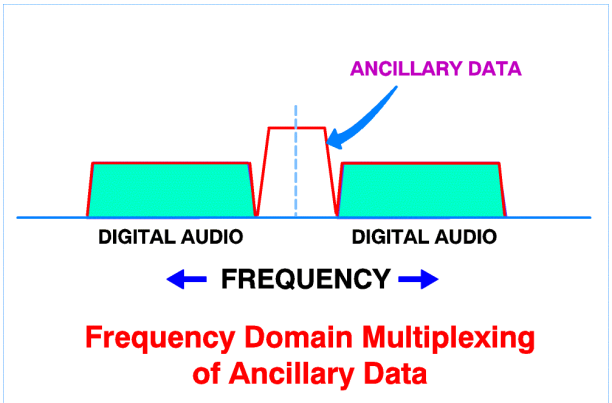

Figure 6

The difference between the two forms of the BPSK ancillary data channel is that while one transmits in bursts at a high data rate, the other transmits slowly but continuously.

In either case, the ancillary data demodulator would also serve as a carrier reference (digital phase locked loop or DPLL) for whatever demodulation method is used by the receiver.

SINGLE CARRIER ANCILLARY DATA

In the time domain multiplexed single carrier systems, a short burst of BPSK carrier would be transmitted at either 0 degrees or 180 degrees with respect to the unmodulated carrier. A single bit could be transmitted in a 125 microsecond time interval.

The overall broadcast signal would transmit its normal single carrier modulation system most of the time, and would occasionally transmit short bursts of ancillary data. For example, 128 bits of ancillary data could be transmitted in an interval of 16 milliseconds. Then 384 milliseconds of compressed audio data could be transmitted, and the cycle would repeat. This would result in an ancillary data rate of 320 bits per second, using 4% of the available time in the broadcast signal.

Each ancillary data packet would consist of a BPSK header, a preamble, one or more subpackets, and FEC overhead. Subpackets contain station identification, decoding method, decoding software, etc. Each subpacket would, in turn, carry a header telling what kind of packet it is, a byte count, and FEC overhead. Ancillary data packets could vary in length and in frequency. So, if a raw data rate higher (or lower) than 320 bits per second is required, it can be obtained simply by changing the packet length and/or the packet frequency.

MULTIPLE CARRIER ANCILLARY DATA

In a multiple carrier system, a BPSK signal would still be transmitted at the station's carrier frequency. However, instead of occupying a percentage of the transmission time, it would occupy a percentage of the spectrum. The multiple carrier ancillary data channel would consist of a signal of approximately 320 Hz bandwidth. Using simple BPSK modulation, this could provide a raw bit rate of 320 bits per second, with the duration of each symbol being 3.125 milliseconds.

OFF AIR DOWNLOADS OF DSP SOFTWARE

Although the architecture and processing requirements of a general purpose DSP receiver are not yet defined, we can make estimates of software download time based on some assumptions.

Assume that the receiver's DSP capability is 100 million instructions per second (MIPs), and that there is one 16 bit word per processor instruction, and that the intermediate frequency (IF) signal is sampled at 44.1 kHz to allow a 20 kHz wide signal to be demodulated. Also assume that the DSP code is entirely "in-line;"

that is, there are no loops, and there is no branching (this will be close to worst case). Using these assumptions, the DSP will be able to perform 2268 instructions for each 44.1 kHz input sample. The DSP software would occupy 16*2268=36288 bits (4536 bytes) of program memory. At a net data rate of 256 bits/sec, this program could be downloaded to the receiver in about two minutes and 22 seconds. Additional time would be required for non-program data such as any DSP filter coefficients, and for packet overhead. The point of this illustration is that download time will be measured in minutes rather than seconds or hours.

If a receiver were to contain 256k bytes of NVRAM for decoding software, it would be able to hold 57 different programs of this size.

At least in the case of the single carrier system, more of the channel capacity could be allocated to ancillary data, which would provide for faster software downloads when a new system is being introduced.

PROPOSED ANCILLARY DATA STRUCTURE

Regardless of whether the ancillary data channel is present in bursts for single carrier or continuously for multiple carrier modes, the proposed ancillary structure is the same (Figure 7).

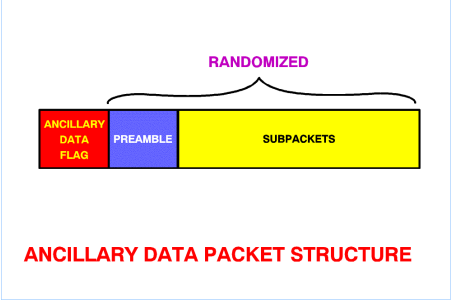

Figure 7

The ancillary data structure proposal is conventional, using several established techniques. It includes Reed-Solomon forward error correction, interleaving, and data randomization.

Each group of ancillary data bits is called a packet. Each packet, in turn, can contain several subpackets. The subpackets contain data such as decoding software, station identification, etc.

The ancillary data packets are BPSK coded, with a 0 corresponding to 0 degrees and a 1 corresponding to 180 degrees with respect to unmodulated carrier.

Each ancillary data packet begins with an ancillary data flag which has three components (Figure 8).

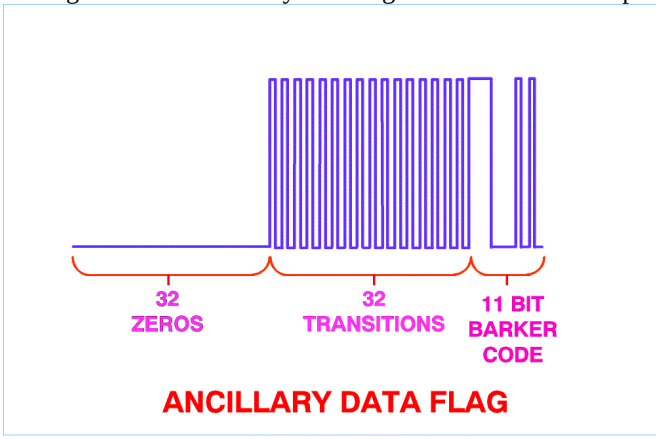

Figure 8

To aid in receiver PLL locking, each ancillary data flag begins with a 32 bit burst of carrier at 0 degrees. In other words, ancillary data begins with a string of 32 zeros. To aid in the locking of the bit clock, the next 32 bits are an alternating 101010... pattern. The final component of the ancillary data flag is an 11 bit Barker code, which assures proper byte alignment[8]. A Barker code minimizes the degree of agreement for shifted versions of the frame pattern. The particular 11 bit Barker code is 11100010010.

The rest of the data following the ancillary data flag is randomized with a pseudorandom sequence (Figure 9). The polynomial for this sequence is 1+X6+X7. This same randomization sequence is used in the SONET (Synchronous Optical Network) telecom standard. Although this sequence repeats after only 127 bits, the maximum run length is only 7 bits.

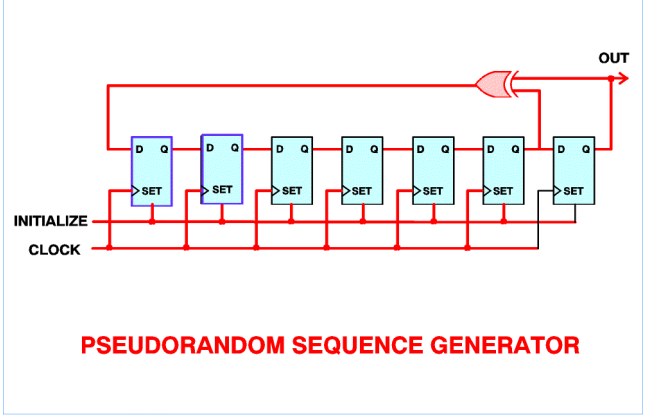

Figure 9

Following the ancillary data flag is a preamble (Figure 10). The preamble is four bytes (32 bits) long. The preamble is coded with a Hamming (31,26) linear error correcting code[9]. 26 bits are data and 5 bits are used for error correction. This code will correct single bit errors and detect double bit errors. The remaining bit is a simple even parity bit taken over all four bytes.

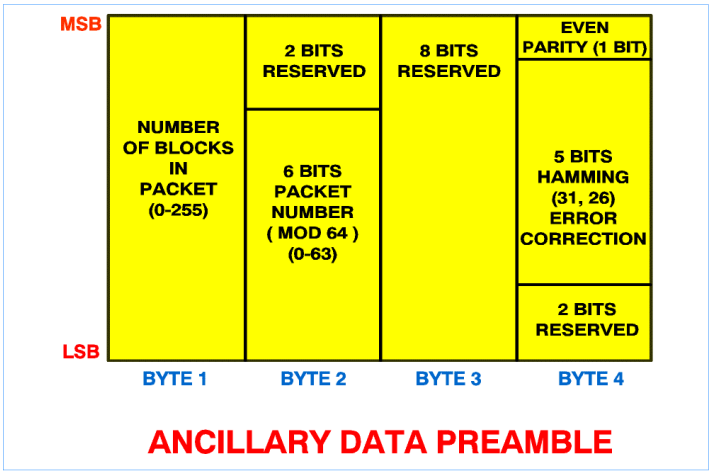

Figure 9

These bytes are transmitted least significant bit (LSB) (bit 0) first. Byte 1 contains an 8 bit integer which carries the number of blocks in the data portion of the packet. A block consists of 15 bytes; this number comes from a conveniently sized Reed-Solomon error correction code. If the data portion of the packet does not exactly fit within an integer number of blocks, the remainder of the block payload will be filled with zeros.

The second byte in the preamble contains a 6 bit value in its 6 LSBs which is a modulo 64 value for the ancillary data packet number. The purpose of this value is to allow the receiver to determine whether it has missed any ancillary data packets. The packets will be numbered 0, 1, 2,... 62, 63, 0, 1,...

The 2 most significant bits (MSBs) (bit 7) of the second preamble byte, along with all 8 bits in the third byte and the 2 LSBs in the fourth byte, are all reserved and should be set to zero. Bits 1-5 of the fourth byte are the Hamming code error correction bits. The LSB (bit 7) of the fourth byte is simple even parity for all four bytes of the preamble.

Subpackets occupy the remaining data space in the ancillary data packets (Figure 11). There are several different kinds of subpackets, containing station identification, receiver software, and other information.

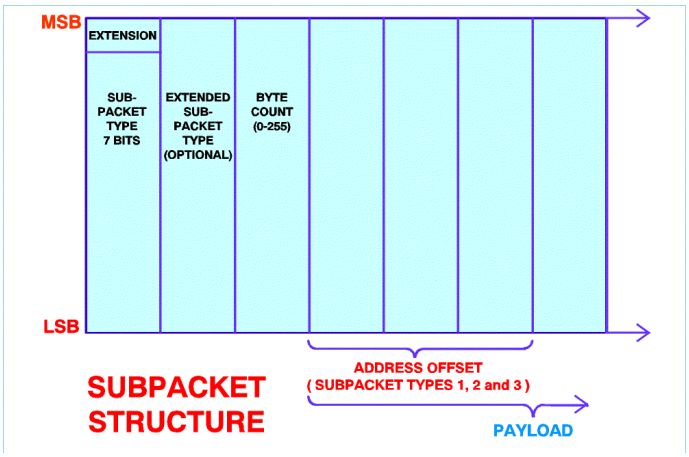

Figure 11

The data transmitted in the subpackets is bit-interleaved (Figure 12). The purpose of the interleaving is to transform burst errors into more correctable isolated errors within the data blocks. The level of interleaving is equal to twice the number of data blocks indicated in the first byte, because each block consists of two sub-blocks. For example, if the ancillary data packet contains 5 data blocks (75 bytes), then the LSB of the first byte is transmitted in sub-block #1A, bit 1 is transmitted in sub-block #1B, bit 2 is transmitted in sub-block #2A, etc. until bit 10 is transmitted in sub-block #1A again. The sequence repeats through the end of the packet.

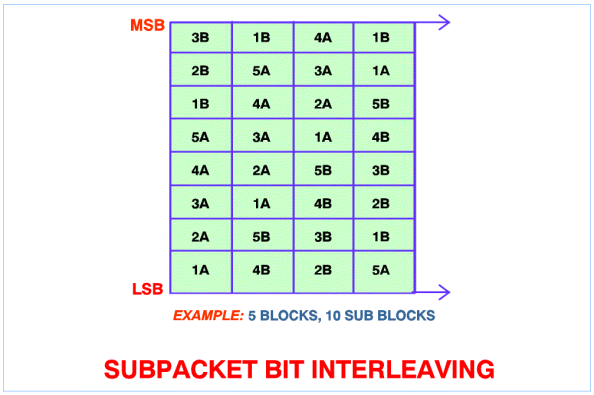

Figure 12

Each block consists of two sub-blocks (Figure 13), each coded with a Reed-Solomon expurgated (15,12) code which operates on a word size of 4 bits[10]. This code will correct any number of errored bits in a single word, and will detect, but not correct, two errored words. Each sub-block consists of 15 half bytes, and one block consists of 15 bytes.

SUBPACKETS

Each subpacket begins with a byte which indicates the subpacket type and its contents. The MSB of this byte is used to extend, if necessary, the subpacket type to the next byte. A single byte can indicate 127 subpacket types, with the MSB=0. If the MSB=1, then the subpacket type extends to the next byte, allowing an additional 32385 subpacket types. It is anticipated that subpacket types will be assigned through standards organizations and/or regulatory agencies.

Following the subpacket type byte(s), the next byte contains a byte count for the remainder of the subpacket. Therefore subpackets can carry a payload of 0-255 bytes.

SUBPACKET TYPES

Only a small number of subpacket types needs to be defined at the outset. When it leaves the factory, the receiver does not need to know about all 32512 possible packet types. The receiver's operating system software itself can be updated as necessary (self-modifying code) through on-air software updates.

Subpacket type 0 is for station identification. It will contain
the following:

a. modulation mode
b. audio compression mode
c. callsign

Optionally, it may also include:

d. station name (e.g. "Newsradio 1530")
e. station location
f. station format
g. emergency information
h. other ASCII messages, including:
    1. Traffic information
    2. Weather forecast
    3. Weather conditions
    4. Program schedules
    5. Time of day
    6. Advertising
    7. Music title & artist

The first two bytes are integers which are indicate modulation mode (0-255) and compression mode (0-255). The remainder of the information in this packet consists of strings of ASCII data. Each of the remaining data types, beginning with the call letters, is preceded by a byte which indicates the number of ASCII characters which follow (0-255). If the value is 0, then the string is not transmitted. For instance, to transmit "KFBK" in the call letters string, the bytes transmitted would be 04h
(byte count), 4Bh, 46h, 42h, 4Bh ("KFBK").

Packet 0 may be terminated early simply by omitting data after the callsign. The optional ASCII strings which follow must be transmitted in the order indicated. If an optional string is to
be transmitted, then all string types which precede it must also be transmitted, although they may be null strings (zero length).

Subpackets type 1, 2, and 3 are used to transmit operating system software for the radio, demodulation software, and audio decompression software respectively.

The first byte following the subpacket type contains a version number for operating system and decoding mode for demodulation and decompression software.

The next three bytes of these subpacket types contain an address offset for the machine code instruction data which follows. The words which follow are written into the radio's nonvolatile program memory starting at the offset address contained in these three bytes.

RECEIVER ARCHITECTURE

The architecture of the DSP portion of the receiver is beyond the scope of this paper. However, its computational power, RAM, NVRAM, instruction set, and other features will be determined by the requirements of present and expected future developments in modulation coding and audio compression algorithms.

A single architecture for a receiver's general purpose DSP would be the most straightforward way to implement a software based system. However, if more flexibility is required, it is possible to allow receiver manufacturers to use different DSP chips with different instruction sets. Such an approach could be implemented by changing the system architecture to transmit some form of abbreviated "source code" to the receiver, which would then compile it to run on the processor actually used in the receiver. Each receiver would include a compiler targeted to whatever DSP chip it contains.

IN-BAND ON-CHANNEL

Some proposed digital AM systems are hybrid analog and digital systems. The signal consists of an analog envelope modulated signal plus some digital modulation which is mostly concealed from the analog receivers. This software based system could be extended to include such in-band on-channel (IBOC) systems.

At least three methods could accommodate IBOC.

First, the ancillary data channel could either phase or preferably quadrature modulate the carrier of the "analog" part of the signal. The data rate would be slow, similar to that of the multiple carrier ancillary data channel.

Secondly, the continuously present multiple carrier version of the ancillary data channel could be moved some 5 to 9 kHz off the center carrier frequency to make room for the analog signal.

Finally, a different ancillary data modulation method could be used which would distribute its energy throughout the channel: spread spectrum.

POSSIBLE FUTURE ENHANCEMENTS

Although this approach would define a standard receiver, there would be nothing that would prevent the development of more advanced receiver types. If additional receiver types are specified in the future, then all that would be necessary would be to transmit several different versions of the decoding software. Each receiver would ignore the subpackets carrying decoding software to other receiver types.

CONCLUSION

A method has been shown which can provide a high degree of flexibility for digital AM. As new modulation and audio compression systems are developed, they can be used almost immediately. Broadcasters will be free to select the transmission methods which best suit their purposes and the needs of their listeners. Different methods can be used for LW, MW, SW, daytime, and nighttime use, with different tradeoffs.

As new modulation and audio compression algorithms are developed, they can be almost immediately applied.

Modern DSP hardware and software technology makes automatically reprogrammable receivers feasible. Use of this technology will allow digital AM to avoid an otherwise high risk of early obsolescence.

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