Digital Audio Appl i cations to Short Wave Broadcasting

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Abstract **- Digital audio is becoming prevalent nut only in consumer electronics, hut also in different broadcasting media. Terrestrial analog audio broadcasting in the AM and FM hands will be eventually he replaced by digital systems. in the United States, developments include in hand digital systems for the FM and AM hands, while in Europe and elsewhere, the Eureka 147 system is undergoing large scale trials. Digital audio broadcasting via satellite is a reality in conjunction with direct to home TV broadcasting. Satellite delivery of audio to mobile receivers is under development. Worldspace has proposed a system of moderate rate digital audio delivery via satellite internationally via a three satellite system, while two organizations have been granted licenses to provide CD quality satellite audio domestically in the 11.S. The one broadcasting service which has indicated an interest in switching to digital, hut is somewhat behind in system development is short wave broadcasting.**

The minimization of data rate is very important in broadcasting because uf bandwidth and power constraints. Audiu compression is therefore used to reduce the data rate required to transmit and reproduce the audio at the receiver. The data rate that is needed depends on the andiu handwidth that is to be maintained, whether it is monaural or stereo, and the distortion level that can he tolerated by the audience. A great deal of effort has heen expended to bring down the data rate needed to reproduce CD quality audio, on the high end of the audio compression performance scale, and on speech compression fur telephony, on the low end uf the scale. Less work has been done in the medium bandwidth speech and music area, which is most appropriate for short wave broadcasting.

The hand width uf a short wave channel is 10 kHz. The narrow handwidth and the very difficult propagation characteristics uf the HF hands will severely limit the data rate that can be transmitted. It will take a very efficient compression scheme to achieve the desired voice and music quality. in addition, attention must he paid to the interaction nf the receiver and audio decoder in minimizing undesirable audio artifacts when the received signal approaches threshold. This paper describes the unique requirements uf audio compression systems suitable for HF broadcasting from the perspective of the overall short wave broadcasting system design.

1. INTRODUCTION

AM broadcasting is one of the oldest broadcasting services in existence. It is most extensively used in the medium wave bands (around I M Hz) for local broadcasting, and in the short wave or HF (3 MHz to 30 MHz) bands for international broadcasting. The HF bands arc suitable for long distance broadcasting because radio signals at these frequencies are reflected from the ionosphere as well as from the surface of the Earth. This allows signals to travel thousands of kilometers by means of one, two, or more hops.

Unfortunately the ionosphere is not an ideal reflecting surface. Its reflective properties vary with time of day, solar cycle, and other seemingly random phenomena, which results in a variety 01 propagation impairments. Listening to a HF broadcast can range from the annoying to almost impossible because of the signal fades, noise, and other impairments, including interference from other stations on the same frequency. In spite 01 this, HF broadcasting has until recently been the best way to disseminate information over a wide area. Now other means of information" delivery, such as the Internet and direct to home satellite broadcasting arc becoming more available. To compete with these, HF broadcasting reception reliability and audio quality will have to be significantly improved. The broadcasting infrastructure is there. A successful conversion to digital broadcasting would go a long way toward improving the quality of short wave audio and returning listeners to this service.

The short wave bands at-c also used by radio amateurs for communications, also using AM. They must also be kept in mind in the conversion to digital process. While there is very little distinction between broadcasting and communications when using AM, the processing delay factor in digital systems can be more of a hindrance in two way communications.

Several independent studies arc currently under way to define systems for HF digital broadcasting. It is hoped to select the best approach and define a "world standard" that most broadcasters will use. A group of international broadcasters and representatives from broadcasting equipment manufacturers arc working to set up a consortium called Digital Radio Mondiale. The goal of this consortium wi I I be to define system requirements and work to facilitate the conversion of HF broadcasting to digital.

One system for digital HF is being studied at the Jet Propulsion Laboratory, under sponsorship" by the Voice of America. This system is being adapted as far as possible from a design that was developed for satellite digital audio broadcasting. Since the audio bandwidths and propagation are di fferent, some of the system trade-oft's have to be redone. Some of the link impairment mitigation techniques that wet-c developed for the satellite system, however, arc applicable and will be useful in a HF broadcasting system design.

II. SIGNAL DESIGN CONSIDERATIONS

A short wave broadcasting channel is limited to a bandwidth of 10 kHz, so the combination of audio data rate and any overhead for synchronization and error protection must fit into this bandwidth. In addition, it will probably be desired by most broadcasters to transmit some auxiliary data together with the audio.

To achieve good audio quality reliably will require a careful set of trade-off's between audio compression, modulation, and coding. It goes without saying that the audio compression must be made as efficient as possible.

A. Audio compression

Audio compression is the means for lowering data rate after the audio is digitized with an A to D converter. Sampling first must be done at twice the audio bandwidth at up to 16 bits per sample. These samples are then processed by the chosen compression technique. There is a trade-off between the final data rate and audio quality. Current state of the art compression techniques which are designed to work with telephone type (3.5 kHz) bandwidths reproduce speech and music fairly well at data rates around 16 kbps. Other techniques which were originally applied to compression of CD quality stereo are being brought down in data rate and can currently handle 9 kHz stereo at approximately 64 kbps.

The audio bandwidth requirement of current AM broadcasting is 4.5 kHz. The initial quality requirement for HF broadcasting should initially provide faithful reproduction of' at least this bandwidth. There is a desire by broadcasters to eventually push this higher. A first cut of the requirements, as proposed by a group of international broadcasters, is shown in Table I.

TABLE 1
AUDIO BANDWIDTH REQUIREMENTS

REQUIREMENT	Short Term	Longer Term
Channels	Mono"	Stereo
Audio Bandwidth	4.5 kHz	9 kHz
ITU-R listening scale	3.5	4.0

It is not yet clear what reliably operational data rate will be achievable over the HF channel, or whether the bandwidth limitation will be eventually relaxed. Progress is being made in defining a robust signal structure, hut at this time it appears that it will be difficult to achieve greater than a 32 kbps data throughput under the best propagation conditions. Backing off to a lower data rate may be required when propagation is less than ideal. Normally this would be done at the transmitter, whenever it is recognized that sufficient signal to noise ratio is not available over the intended coverage area.

B. Auxiliary Data

In digital transmission systems it is possible and usually desirable to send program related or other data together with the audio. in order to save bandwidth, non time critical data can be combined with the compressed audio signal, taking advantage of periods when the information content of the audio is low.

Since audio compression schemes work with finite time segments of audio, there is some time averaging of the information content of the signal. Nevertheless there are times, such as a period of silence, when the information content is very low. To really smooth out the slow variations would require working with long time segments of audio, which would result in a large delay in restoring the audio at the receiver. Thus introducing auxiliary data when audio information is low is a way of utilizing bandwidth more efficiently.

Audio frame synchronization and error protection coding, on the other hand, must be supplied at a fixed data rate. Therefore these must be added as a fixed overhead to the transmitted data stream. The resulting signal, showing the composite elements, is illustrated by Fig. 1.

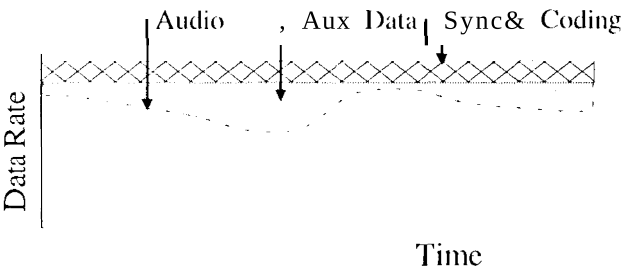

Fig. 1. Components of composite data stream

C. Error Protection

A digital communication channel is subject to bit errors. These bit errors can be bursty or more evenly distributed in time, depending on the characteristics of the channel. Bit error protection is normally done through forward error correction (FEC) coding on the signal, but some error protection is usually also applied as part of the audio compression process. The audio compression system knows which data bits arc more important and can assign more protection to them. Not-really frame sync headers arc more heavily protected than the audio information. However if the audio information bits can be ranked in importance, then the more important ones can be better protected, which should result in more graceful degradation of the audio with increasing bit error rate.

There are error concealment methods which are used to mask the presence of uncorrectable errors. One such is the repeating of the previous frame when a frame is corrupted.

This works for the loss of individual frames, but rapidly fails when multiple frames are corrupted. In the case of stereo, a corrupted frame in one channel can be replaced by the corresponding frame from the other channel. A possible technique might be to error protect the low frequency components of" the signal more than the high frequency components. When errors start to occur, there would initially be a loss of frequency response rather than annoying artifacts.

In digital transmission systems, the difference between a barely noticeable error rate and Vet-y annoying audio decoder performance is a matter of a few dB in signal to noise ratio. Nevertheless, the manner in which the audio degrades as the link starts to fail is an important design issue. When listening to news, for example, the listener may tolerate a higher level of distortion" than when listening to music. If possible, the nature of the audio degradation should be control led, so that errors do not produce loud disturbing noises which can harm the listener's ear or damage the receiver.

It is possible to design the receiving system to completely mute the audio decoder as soon as audio quality degradation becomes barely perceptible. It is also possible to allow the audio decoder to operate until the bit error rate becomes quite high, resulting in very annoying audio under certain signal fade conditions. "1'here is no clear answer yet where the muting threshold should be set in a HF broadcasting system. It will depend on the propagation conditions, which can be highly variable, as well as on listener preference. Thus it may be desirable to allow some listener adjustment of Ibis parameter in the receiver.

D. Modulation and Coding Trade-offs

The 10 kHz bandwidth restriction limits the channel symbol transmission rate to about 8000 symbols per second, assuming suitable pulse shaping of the symbols. 'the audio bandwidth and quality requirements of HF broadcasting will require compressed audio data rates on the order of 16 kbps to 48 kbps. To transmit this over a 8 ksps channel will thus require a modulation scheme capable of 2 to 6 bits per symbol; exclusive of any synchronization or coding overhead. The upper value may only be possible under very benign propagation conditions, as well as with a very careful receiver design.

The HF channel usually suffers from multipath and one of the solutions is equalization, which requires some overhead for training symbols, Given the requirement for this overhead plus some more for synchronization and coding, the maximum audio rate to be expected to work in a 10 kHz channel will be less (baa 32 kbps.

An example modulation and coding trade-off space from which a system design can be constructed is illustrated by Fig. 2, which shows several modulation-FEC coding combinations. Shown arc the relative bandwidth and signal to noise ratio requirements relative to uncoded, pulse shaped BPSK, which has a spectral occupancy of approximately 0.8 bits per Hertz.

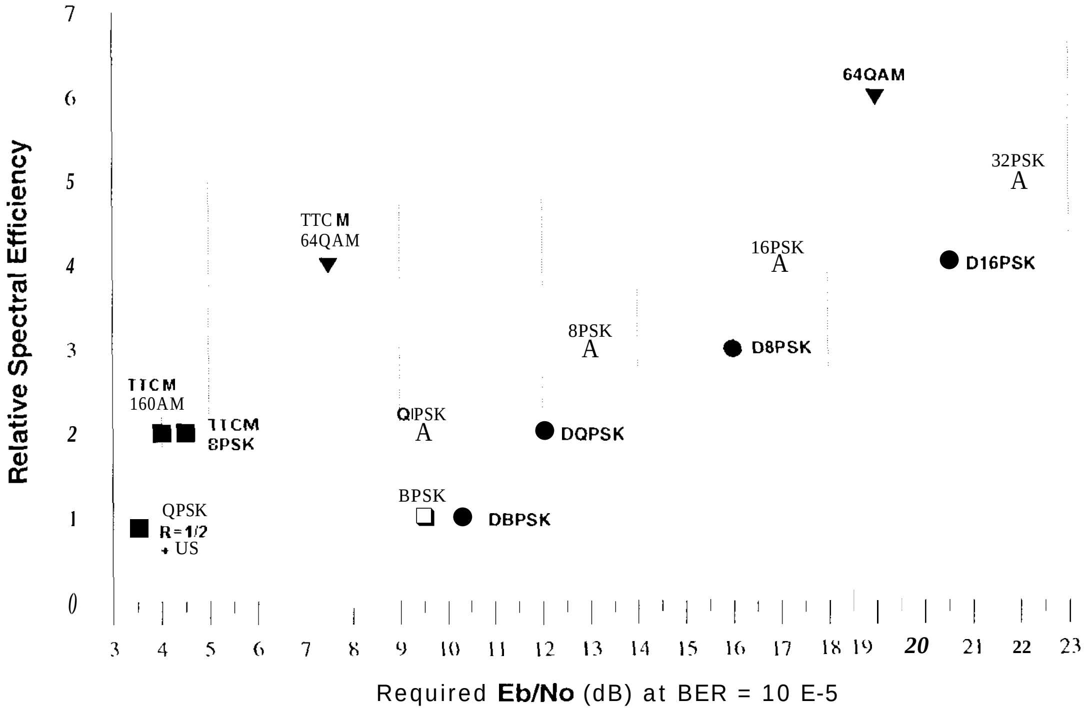

Fig 2. Modulation and coding trade-off space

Higher levels of modulation require higher signal power for an equivalent bit error rate. On the other hand, higher order modulation allows a higher data rate within a given bandwidth, including more room for forward C1"1'or correction (FEC) coding. More powerful FEC codes in turn lower the required signal power, but d ecrease (IIC data rate available for the audio.

The availability 01 transmitter power is a significant factor in the choice of the signal structure. II' bandwidth is more important than power, then an uncoded signal with as high a modulation factor as possible is the proper choice. If transmit power is critical, then FEC coding must be used and the audi o data rate must be lowered.

111. SUMMARY ANI) CONCLUSIONS

The restricted channel bandwidth available for 1 IF broadcasting will require very efficient audio compression, with good quality for music as well as speech. There is insufficient experience to dale with higher order digital modulation over the HF channel to determine the practical data throughput limit, but presently indications are that this limit is below 32 kbps in a I () kHz channel. It is likely that there may be a requirement of variability in compressed audio data rate or oth er parameters to adapt a given broadcast 10 varying transmission channel conditions,"

Clearly there is interaction between the design of the audio compress ion system and the defini (ion of the signal structure. It is important that error mitigation be distributed correctly between the audio decode' and the re ceiv er. The way that the audio behaves near receiver threshold is also a design issue.

The following audio compression system parameters should be kept in mind during the near term development of digital HF broadcasting:

- High quality speech and music at data rates in the range of 24 to 48 kbps (see '1'able I). There will be a strong push to bring (his upper limit down and still maintain audio quality.
- Ability to include multiplex of auxiliary data within the audio data stream.
- Good error protection and concealment.
- User adjustable muting point
- Reasonable audio decoder processing requirements (cost and receiver battery life considerations)

IV. ACKNOWLEDGMENT

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