

ACSB - A MINIMUM PERFORMANCE ASSESSMENT

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The work presented in this paper was supported by the National Institute of Justice of the U.S. Department of Justice as part of their Technology Assessment Program.

ABSTRACT

Amplitude companded sideband (ACSB) is a new modulation technique that uses a much smaller channel width than does conventional frequency modulation (FM). Among the requirements of a mobile communications system is adequate speech intelligibility. This paper explores this aspect of "minimum required performance." First, the basic principles of ACSB are described, with emphasis on those features that affect speech quality. Second, the appropriate performance measures for ACSB are reviewed. Third, a subjective voice quality scoring method is used to determine the values of the performance measures that equate to the minimum level of intelligibility. It is assumed that the intelligibility of an FM system operating at 12 dB SINAD represents that minimum. It was determined that ACSB operating at 12 dB SINAD with an audio-to-pilot ratio of 10 dB provides approximately the same intelligibility as FM operating at 12 dB SINAD.

INTRODUCTION

The underlying impetus for this work is the growing crowding of the frequency spectrum for mobile radio users. Since the use of less bandwidth per channel is an apparently obvious way to reduce crowding, there has been increased interest in recent years in "narrowband technologies." The advantages of narrowband techniques are of considerable importance in the Mobile Satellite Service (MSS) as well as the land mobile services, since maximum utilization must be made of a limited spectrum allocation. Both analog and digital techniques

have been explored for possible use in MSS applications. ACSB is an analog technique that requires only a fraction of the channel width required by frequency modulation (FM). While a narrow channel width may contribute to spectrum efficiency, a wide channel width may contribute to voice quality. The goal, therefore, has been to find a modulation scheme that increases spectrum efficiency and retains the voice-quality level that mobile radio users require.

The purpose of the work reported herein is to identify the appropriate performance measures for ACSB, and to determine the values of such measures that represent the minimum level of acceptable performance. Many aspects of performance were considered. These include spectrum efficiency, propagation range, application flexibility, noise susceptibility, and voice quality. In this paper some of these are discussed, but the focus is on voice quality. The minimum level of acceptable voice-quality performance for ACSB is chosen to be the same as the minimum level of acceptable performance for FM, and the quantifiable aspect of voice quality used to represent this performance is chosen to be intelligibility as represented by the articulation score (AS).

In what follows, a brief technological description of ACSB is given and the applicable voice quality measures are described. Last, the measurements made to determine the values of the chosen performance measures are described.

HISTORY OF ACSB

Early research and development of ACSB was sponsored by the Federal Communications Commission. Much of this early work was done by Bruce Lusignan and others at the Communication Satellite Planning Center at Stanford University. Based on this work, the FCC issued a Notice of Inquiry (NOI) to solicit comments regarding the introduction of narrowband technology into the land mobile bands (FCC, 1980). The responses ranged from strong support for ACSB to claims that the Stanford study was incomplete. Other comments indicated that more investigation was necessary and that the spectrum efficiency was not well defined. In spite of the controversy, it was clear that narrowband technology was a better answer to the need for more land mobile channels than to allocate more spectrum. Following this NOI, the FCC allowed developmental licensing of ACSB stations in the 150 MHz land mobile band.

A Notice of Proposed Rulemaking (NPRM) released later proposed 5-kHz channel assignments in the 150 MHz land mobile band. This NPRM resulted in the adoption of 5 kHz channels with an offset of 2.5 kHz, and amendments were made to Part 90 of the FCC Rules and Regulations (FCC, 1985a). This action authorized permanent licensing (as opposed to developmental licensing) of private land mobile stations using narrowband technology.

FUNCTIONAL DESCRIPTION OF ACSB

ACSB is a form of single-sideband suppressed carrier (SSBSC) modulation. The main advantages of SSBSC are its efficient use of the

frequency spectrum and superior signal-to-noise characteristics compared to conventional double sideband amplitude modulation (AM). Since SSBSC uses only one of the two sidebands generated in AM, the bandwidth required is only the bandwidth of the voice frequencies. Since no carrier is generated in SSBSC, the transmitter has no power output until it is modulated. Also, all power is concentrated in the sideband, where the information is contained. This factor, combined with the narrower bandwidth, leads to superior signal-to-noise characteristics (for the same transmitter power) for the demodulated signal of SSBSC as compared to AM.

While the absence of the carrier contributes to the superior signal-to-noise characteristics, it is a major drawback of SSBSC. Since no carrier is received, one must be reinserted by the receiver in order to demodulate the signal, and if the reinserted carrier is not precisely on frequency, the result will be serious distortion of the demodulated signal. The voice frequencies will be offset in frequency by the amount of the error in the frequency of the carrier that is reinserted. Since ACSB transmits a pilot tone as a frequency reference, this problem is eliminated.

The ACSB equipment currently manufactured in the United States uses a tone-above-band pilot at 3100 Hz. Figure 1a illustrates an idealized frequency spectrum of an ACSB transmitter in the currently accepted format. Figure 1b illustrates the frequency spectrum of the same transmitter, but with a 1-kHz tone audio input.

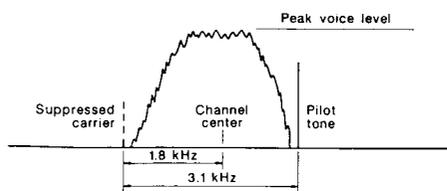


Figure 1a.

Idealized emission spectrum of an ACSB signal illustrating the envelope of voice modulation. The nominal range of the voice band is from 300 to 3,000 Hz.

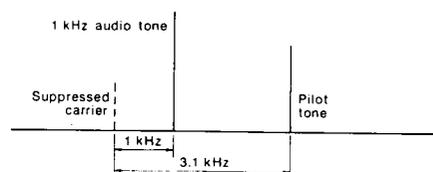


Figure 1b.

Idealized emission spectrum of an ACSB signal modulated with a 1-kHz tone.

Besides providing an accurate frequency reference for the receiver, the pilot tone also may provide:

1. automatic gain control at rf and/or IF,
2. squelch,
3. selective signaling, and
4. expander gain control (feed forward gain control).

ACSB uses syllabic companding in addition to the pilot tone. The term "compand" is a combination of two other words: compress and

expand. A compandor is a device that performs companding functions. It should be noted that the term "compandored" is often used and that it means the same thing as companded. The term companded is used throughout this report. Compression (and pre-emphasis) occurs in the audio circuits of the transmitter and expansion (and de-emphasis) occurs in the audio circuits of the receiver. The term "syllabic companding" implies that the compression and expansion response times are consistent with the syllabic rates in speech. The system currently employed by manufacturers of ACSB land mobile equipment in the United States uses two stages of 2-to-1 compression in the transmitter and two stages of 2-to-1 expansion in the receiver. Figure 2 illustrates this companding with the introduction of noise in the transmission path. The dynamic range of the illustrated signal

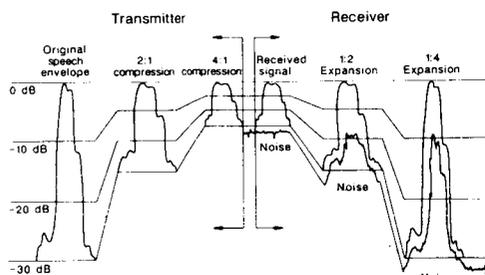


Figure 2. Effect of ACSB Expansion.

(consider it to be a syllable) is 30 dB at the input to the first compressor. The dynamic range of the signal leaving the first compressor is 15 dB. Next, the signal enters the second compressor; the signal leaving the second compressor has a dynamic range of only 7.5 dB--1/4 of that entering the first compressor. The effect of reducing the dynamic range of the signal is to increase significantly the median level.

The received signal-to-noise ratio is improved by the use of compression, and the dynamic range of the audio signal is restored by expansion. Expansion does not improve the signal-to-noise ratio because the instantaneous expander gain must be applied to both the signal and the noise at the same time.

Figures 3 and 4 are simplified block diagrams of an ACSB transmitter and receiver, respectively. The block diagram for the transmitter shows the locations of the two compressors and the injection of the pilot tone before the second compressor. The receiver block diagram shows the locations of the two expanders and the processing of the pilot tone to provide frequency lock, automatic gain control, and expander gain control.

Using a computer model based on these block diagrams, some simple results may be obtained. Figure 5 shows a comparison of the emitted rf power contained in the information sidebands for ACSB and SSBSC. This shows the effect companding has on the output power. Figure 6 shows the relative amounts of rf power contained in the information sideband and the pilot for ACSB.

PERFORMANCE CRITERIA

The ultimate goal in setting specifications for equipment is to determine if the equipment performs as expected by the user. Determining precisely what is expected is not possible for each piece of equipment and every user. Therefore, the specifications are usually given in terms of parameters that are user-oriented either by design

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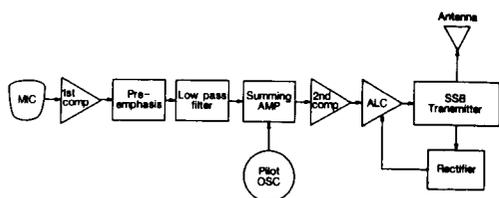


Figure 3.
Simplified ACSB transmitter block
diagram.

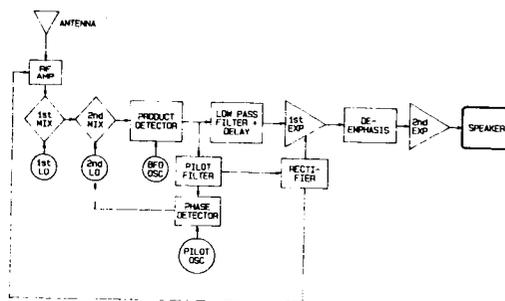


Figure 4.
Simplified ACSB receiver block
diagram.

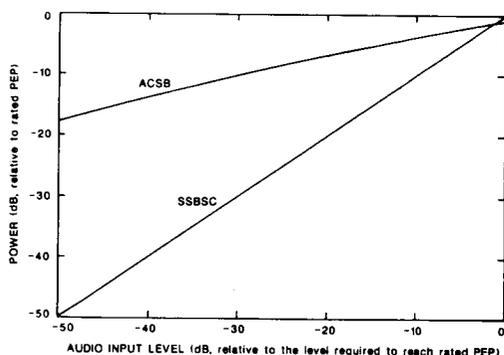


Figure 5.
The rf power contained in the
information sidebands of ACSB and
SSBSC vs. audio input level.

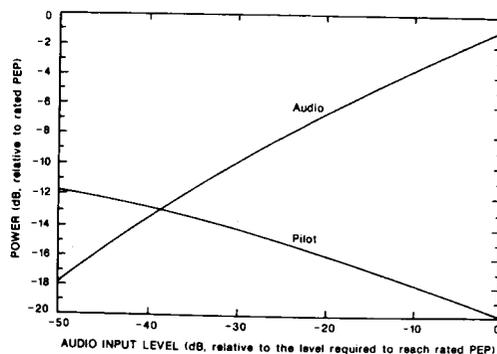


Figure 6.
The rf power contained in the
audio component and the pilot vs.
audio input level.

or by becoming accepted due to a long period of use. In this study, the use of the subjective intelligibility testing procedure using phonetically balanced word lists and trained listener panels is chosen as the user-oriented measure (ANSI, 1960).

The subjective tests are usually too expensive to be used for routine specification testing, so an objective measure that can be related to the subjective tests is required. Here, SINAD is proposed because it includes interference and distortion.

SINAD--the objective measure to be used in these tests--is the acronym for

$$\text{SINAD} = 10 \log \frac{s + n + d}{n + d} \quad \text{dB}$$

where s = signal
n = noise
d = distortion

The relationships between the subjective and objective measures for both ACSB and FM are shown in the illustration of Figure 7. A thorough review of all the subjective and objective measures is not possible in this paper so the discussion is limited to the chosen subjective and objective measures.

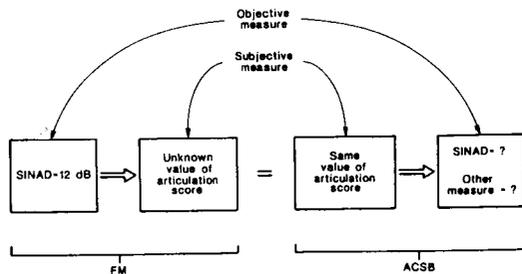


Figure 7.
Relationships between the known and unknown measures for FM and ACSB showing the role of subjective articulation scoring.

It is easy to design a subjective measure for the purpose of specifying the performance of a voice channel. However, the paucity of good designs that include detailed protocol for the testing presents a tremendous obstacle to the understanding and acceptance of voice-performance measures. For our purposes, a subjective measure must be selected that can be related to some objective measure since using subjective measures for acceptance testing has proved to be expensive.

Subjective testing is reviewed in Nesenbergs et al. (1981). The types of tests that have been used are too numerous to be covered here. The two types of testing that have gained credibility are first, the intelligibility tests, and a distant second, the diagnostic tests such as the diagnostic rhyme test (House, 1965).

Since the goal is to determine the acceptability of equipment performance but not the reason that the performance is acceptable or unacceptable, and because of the high costs, the diagnostic tests were eliminated from further consideration. Even though some of the diagnostic tests measure characteristics other than intelligibility, there is little evidence that these measures are useful for predicting user acceptability for systems that use analog modulation. Thus, the relatively simple intelligibility testing using the phonetically balanced (PB) word lists and trained listener panels was used.

The phonetically balanced word lists are divided into word groups of 50 words. These word groups are recorded using several male and female talkers so that one 50-word group represents a cross section of the talkers. The talkers are carefully chosen, usually from radio broadcasters and the recordings are done under studio conditions with the recording level on the tape carefully controlled.

The articulation scoring is done by the U.S. Army Electronic Proving Ground, Ft. Huachuca, AZ. The procedure uses 50-word phonetically balanced word groups, selected and trained listener panels, and computerized analysis of responses. The results are generally accepted as accurate and repeatable and thus suitable for comparison to objective measures (Nesenbergs et al. 1981). Because of the structured nature of the test, and the psychological nature of speech perception, the results may not be conveniently applicable to a user perception of "real life" requirements. However, the testing and

interpretation have displayed a longevity indicating a degree of usefulness that is not easily measured.

In most cases, the signal is a 1-kHz tone which is used as input to the baseband of the transmitter. If the methods of making the measurements are specified, SINAD represents a measurement which is repeatable.

A rather interesting finding is indicated in Figures 8 and 9. Figure 8 shows that at a given audio component signal strength the SINAD of unit A is dependent on the audio-to-pilot ratio. However, in Figure 9 it is seen that for a given audio component signal strength the SINAD of unit B is relatively independent of the audio-to-pilot ratio. This difference in performance apparently results from the use of a more narrow pilot filter in unit B.

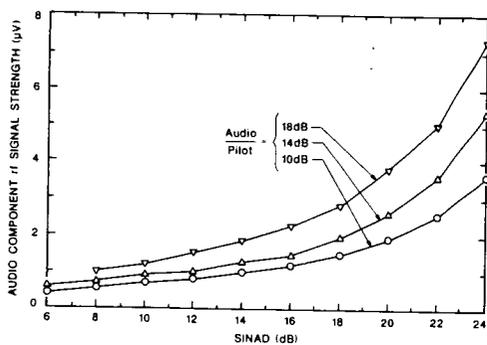


Figure 8.

Measured SINAD vs. audio component rf signal strength for unit A.

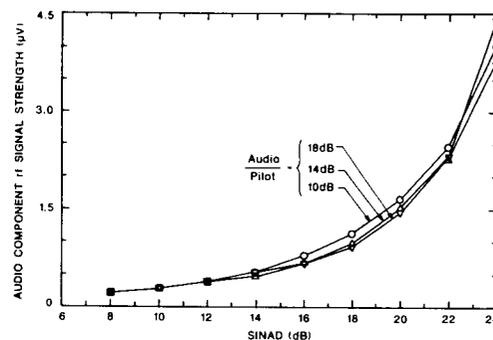


Figure 9.

Measured SINAD vs. audio component rf signal strength for unit B.

CONCLUSIONS

From Figure 8 we see that for some ACSB receivers, SINAD is significantly dependent on the audio-to-pilot ratio. Therefore, the SINAD sensitivity for ACSB radios should be specified at a particular audio-to-pilot ratio. The articulation scores versus SINAD (audio-to-pilot ratio of 10 dB) for the two ACSB radios are plotted on Figure 10. The results for the FM radios are plotted on Figure 10. At 12 dB SINAD the articulation scores for the two FM radios are very nearly the same, and if these are averaged, the result is 88.6%. The articulation scores for the two ACSB radios are not the same, but if they are averaged the result, 88.5%, is very close to the average for the two FM radios. The analysis easily leads to the following conclusion: ACSB operating at 12 dB SINAD with an audio-to-pilot ratio of 10 dB provides very nearly the same intelligibility as FM operating at 12 dB SINAD.

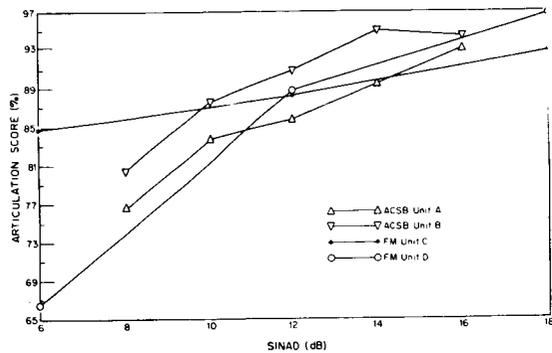


Figure 10.
 Articulation score vs. SINAD for
 ACSB operating at 10 dB audio-to-
 pilot ratio compared to FM.

REFERENCES

- ANSI 1960. Method for measuring monosyllabic word intelligibility, American National Standards Institute, 10 East 40th Street, New York, NY.
- FCC 1980. Notice of Inquiry, PR Docket 80-440, Adopted August 1, 1980, Released September 9, 1980.
- FCC 1985a. Report and Order, PR Docket 84-279, Adopted March 1, 1985, Released March 29, 1985.
- House, A. S., C. Williams, M. H. L. Hecker, and K. D. Kryter, 1965. Psychoacoustic speech tests: a modified rhyme test, ESD-TDR-63-403, Hanscom Field, Bedford, MA.
- Nesenbergs, M., W. J. Hartman, and R. F. Linfield, 1981. Performance parameters for digital and analog services modes, NTIA Report 81-57.