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MSAT Voice Modulation Considerations

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ABSTRACT

The challenge for MSAT voice services is to provide near-toll quality voice to the user, while minimizing the power and bandwidth resources of the satellite.

The options for MSAT voice can be put into one of two groups- Analog and Digital. Analog, nominally narrowband single sideband techniques, have a demonstrated robustness to the fading and shadowing environment. Digital techniques, a combination of low-rate vocoders and bandwidth efficient modems, show the promise of enhanced fidelity, as well as easier networking to the emerging digital world.

The problems and trade-offs to designers are many, especially in the digital case. Processor speed vs. cost and MET power requirements, bits to be allocated to the voice encoding vs. channel coding, bandwidth efficiency vs. power efficiency etc. While the list looks daunting, in fact an acceptable solution is well within our technology.

This paper sets the objectives that the MSAT voice service must meet, and reviews the options that we see now and in the future.

OUTLINE

This paper first addresses voice service on MSAT from the point of view of users to gain an appreciation of the requirements, then from a system designer's viewpoint as he/she tackles the requirements.

USER'S VIEW

The most eager adoptors of MSAT will be people who, because of where they live or work, receive very poor service from existing terrestrial systems, if they get service at all. "Poor service" typically means low availability (does not work everywhere), unreliable (doesn't work when he wants it), low quality (cannot understand or recognise the talker).

MSAT is a natural service to satisfy the availability requirement, since it will provide coverage of all of North America. MSAT will also provide a very reliable link, compared to HF communications which are the only available in many areas. Although the link will suffer when shadowed with trees, the user has some control since he can move to a clear location. Unfortunately, he cannot change the atmosphere to create a better HF channel.

Voice quality is the most nebulous, hardest-to-get-a-grip-on issue facing the design of the voice service, because there are so many variables that affect quality, and the degree to which it affects quality is different for each person. Furthermore, some customers will accept any quality as long as the voice is intelligible, while others are more discriminating. Most users will ask for "near-toll quality".

SYSTEM'S VIEW

The system designer looks first at the user needs, then factors in space and ground segment costs, the propagation channel, modulation, available technologies, coding and voice encoding; then attempts to come up with the most cost effective solution. All the above factors interact with each other, making the solution difficult. Given that we cannot objectively quantify the interaction makes finding an optimal solution nearly impossible.

Defining the problem

The goal of any non, non-profit company is to make an acceptable return on investment. TMI's business plan calls for providing a high quality service at a modest user cost.

The cost to TMI is dependent on the cost of network (satellite and earth stations) and the number of channels, which in turn is dependant on the total resources of the satellite (bandwidth and power) and the resources required per channel. But a factor in the per channel power and bandwidth is the antenna gain and the quality the channel is to provide. Generally speaking, a more complex ground segment will reduce the resource requirements, while increasing the quality will use more resources.

In addition to these technical trade-offs, we must also take into account the behaviour of the users. Customers will naturally pay more for higher quality. But how much more? The "trick" is to find the area where the value to customer is greater than the service provider's cost by a sufficient margin to ensure a reasonable profit. TMI believe this margin does not lie in providing a "low budget" service.

Developing solutions

The first trade-off is essentially one of mobile antenna gain vs satellite EIRP. Potential users are sensitive to the upfront cost of a MET and, therefore, one is not likely to specify a high gain (i.e. cost) antenna. The exact value of gain is still under discussion, however, a 10 to 12 dBic gain is thought to be reasonable.

In Figure 1, satellite capacity in assignable channels is related to MET antenna gain. At 10 dBic gain, the capacity of the system is about 1480 assignable channels.

More complicated trade-offs are dependent on the modulation. Therefore, we will examine potential analog and digital solutions to voice transmissions.

Analog solution. The capacity example of Figure 1 is derived assuming Amplitude Companded Single Sideband (ACSSB). ACSSB, described in [1], is an analog modulation technique that requires only 5 kHz of RF bandwidth. The demodulator makes use of a pilot tone to deal with the propagation effects. The result is a highly robust signal that can tolerate fades of 12 to 14 dB while maintaining intelligible voice.

Figure 2, adapted from the results in [2], depicts the variation of Mean Opinion Score, MOS, with carrier-to -noise density ratio, C/No. In the MOS method, speech samples are played for a large population of evaluators who are asked to rate the quality from 1-BAD to 5-EXCELLENT. The average of each person's rating is the mean opinion score.

The figure shows that quality peaks at about 51-52 dB-Hz with a rating of about 4.1 (GOOD). It also shows a graceful degradation down to a MOS of 2 (POOR) at about 42 dB-Hz. Recent improvements to ACSSB promise to shave 3 to 5 dB off the necessary power.

Because ACSSB is a linear modulation technique, a linear amplifier is required. The peak power of the signal is approximately 6 dB above the average power.

ACSSB also provides a transparent channel that allows direct operation of facsimile, telephone modems, and DTMF over-dialling. From this point of view, ACSSB is more natural for interfacing to the PSTN.

In order to save satellite power, the North American MSAT system will employ voice activation. Figure 3 shows the effect of various levels of voice activation on satellite capacity. The residual pilot level is level of the pilot, in dB, during speech pauses, relative to the power in the signal during speech (including pilot). During speech, approximately half the power of the signal is in the pilot. Thus a pilot level of -3 dBc, indicates no voice activation and the capacity drops by over 40% compared to total voice activation. Even at pilot levels 12 dB down, the capacity drops by about 8%.

Although these calculations were performed assuming ACSSB, digital voice would experience the similar problems if the carrier remained on during speech pauses. The difference is that higher levels of voice activation should be easier to implement with ACSSB than with digital transmissions. This is another point in favour of ACSSB.

Digital solution. In the discussion of ACSSB above, we saw that while ACSSB is very robust in a faded environment, it requires 51 - 52 dB-Hz to achieve good quality in a static environment. Thus ACSSB is not the best choice in uses where fading and shadowing are relatively light.

The trade-off between digital voice quality and satellite resources is not so easy. In a digital system, resources of the satellite are used to pass bits between users. (Generally speaking, the more bits/sec that the satellite passes, the higher the quality. However, quality does not improve indefinitely, but there is a point beyond which increasing the rate does not buy better quality. What increasing the bit rate can buy is reduced complexity.)

Conventional transmission of digital voice exhibits a threshold effect. When the C/N ratio drops below a certain point, the quality quickly deteriorates. Thus in fading and shadowing, ordinary digital systems are weak. However, digital voice has some properties that can be exploited to over come this threshold to some extent. Error control coding can be applied to specific bits that are most sensitive to errors. This can raise threshold BER performance from about 10^{-3} to 10^{-2} with a minimum of coding overhead. Furthermore, techniques such as frame repeat, or frame substitution can be used to fill in for badly corrupted frames. These techniques can be used to overcome bursts errors of 60 - 90 ms in length.

The issue of modulation schemes for the mobile environment is one which has had considerable study. INMARSAT and AUSSAT have both specified QPSK. The Jet Propulsion Laboratory have developed an 8-PSK Trellis Coded Modulation (TCM) modem as part of their MSAT-X work for NASA [6], and the Communications Research Centre are presently developing a 16-QAM TCM modem [7].

QPSK is relatively easy to implement, and Offset-QPSK, INMARSAT's choice, has a near-constant envelope which reduces the linearity requirement on the amplifier. However, compared to TCM schemes, this modulation is not bandwidth efficient. Since the first generation AUSSAT and INMARSAT satellites will be power limited, QPSK is a logical choice.

The MSAT satellites, on the other hand, will be dedicated high power satellites and may be both power and bandwidth limited. Therefore a power and bandwidth efficient modulation scheme is required.

Voice quality testing. For the testing of digital voice, ideally we would take all likely combinations of vocoders, background noise conditions, modulation, coding, propagation conditions and interleaving, measure the quality of each combination, weight each by the cost (satellite resources) and the probability of occurrence of the impairments (background acoustic noise and propagation conditions). At the end, we would have the best design for the voice service.

Testing the quality of analog voice is considerably simpler, since the voice "coding" and modulation is one process. This eliminates the all the vocoder/modulator combinations.

The real problem is measuring the quality of each combination. Objective measurements such as signal-to-noise ratio can be used. But this can't account very well for the physiological aspects of human hearing. A more accepted solution is the subjective Mean Opinion Score.

MOS is not without its problems either. It is a one-way test; this is especially significant because of the variable (from one design to another) delay of the encoding and interleaving process, on top of the fixed satellite delay. The test cannot judge how people would rate the quality in an actual conversation.

A bigger drawback is that full factorial testing of all the factors is clearly not practical. To reduce the number of tests for the digital solution, it is common to separate the modulation and channel coding from the voice encoding. The effects of the channel are therefore manifested in terms of bit errors. The task is often further subdivided by separately testing the effects of uniform errors, burst errors and background noise environments.

This is the procedure of the vocoder testing that Telecom Australia performed on behalf of INMARSAT and AUSSAT [3,4], and in-house testing at Telesat Mobile [5].

To compare analog and digital solutions, the vocoder must be matched to an RF modem, and tests made over a propagation simulator. TMI and the Communications Research Centre intend to perform such tests when CRC develops the necessary hardware this summer.

SUMMARY

TMI and AMSC have specified flexibility in the design of the ground segment for the MSAT system. Vocoders and RF modems will continue to improve with time, and it is the objective for the system to take advantage of these improvements.

The MSAT signalling system will allow up to 16 different voice coding/modulation channel types. By taking advantage of Digital Signal Processors, we hope to minimize the amount of hardware that is necessary to support the different types. Wherever economic, DSP software will be used to replace hardware.

TMI will be monitoring closely the vocoder testing in Australia, and the TCM work at CRC. In the fall of 1990, we hope to be able to test the integration of vocoder(s) and the TCM modem over a propagation simulator. We will also test improved versions of ACSSB, which will also be available.

It is expected that comparison of the two will show that ACSSB is superior in moderate-to-heavy shadowing that many customers may experience in Canada due to the low elevation angles. Other customers, who are not using the system in harsh environments may be able to take advantage of a higher fidelity digital system.

The table below summarizes TMI's current view of the analog and digital solutions. Each system has its advantages over the other in different areas, as indicated by check marks. For this reason, we plan to support both ACSSB and a digital scheme when MSAT is operational in 1994.

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Comparison of Analog and Digital Solutions

	Bandwidth	Interface to POTS	Interface to Digital Networks	Unfaded Quality	Heavily Shadowed Quality	Potential for Improvement
Analog	✓	✓			✓	
Digital			✓	✓		✓✓

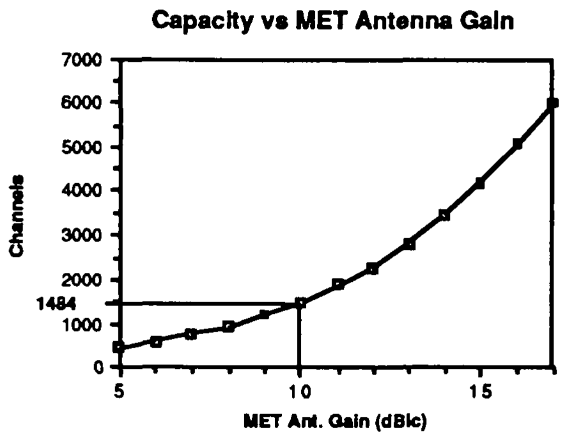

Figure 1

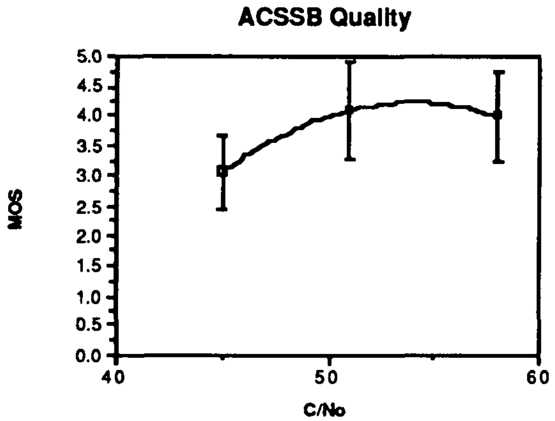

Figure 2. ACSSB Quality (C/No is in dB-Hz)

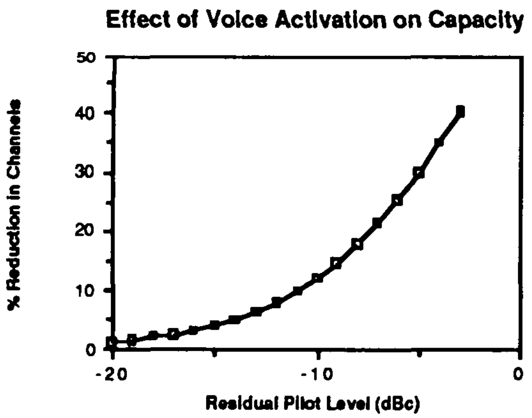

Figure 3