INTRODUCTORY SYSTEM DESIGN OF THE ADVANCED SUBMINIATURE TELEMETRY SYSTEM

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ABSTRACT

The Advanced SubMiniature Telemetry System (ASMT) with Wireless Sensor extension is an ambitious program aimed at incorporating modern wireless system and electronic design methods into a two way, miniature, low cost, modular, and completely software controlled wireless data acquisition system. The program was conceived and is sponsored by the U.S. Air Force SEEK EAGLE Office as a means of both lowering test cost and increasing test effectiveness. This article shall present the fundamental system design challenges of the program and how modern design methods can provide a new standard of cost effectiveness, mission capability, and high spectral efficiency.

KEY WORDS

Telemetry, Wireless, Data Acquisition, Airborne Testing, Spread Spectrum

INTRODUCTION

The Air Force SEEK EAGLE Office (AFSEO) identified the need in the early 1990's for a highly compact, low cost, sensor, data acquisition, and radio telemetry system that could be easily used on general line aircraft. It was envisioned to use a "Peel and Stick" method of hardware mounting that allowed easy removal for use on other aircraft or stores and quick return of line aircraft to standard duties. An "electro-cleavable" adhesive has been successfully developed by EIC Laboratories (www.eiclabs.com) for this function. Hardware cost was desired to be sufficiently low as to allow sacrifice in destructive testing when necessary. This methodology avoids the need to always use expensive, low availability, pre-configured test aircraft for all testing. Not only could this concept lower test expenses, it could also result in better test programs by eliminating aircraft availability delays and allowing more testing per mission. In this same period of time, Cleveland Medical Devices (www.clevemed.com) has been pursuing development of wireless patient monitoring systems based on shrinking cellular methods in physical size and power consumption down to the point that they become "wearable". We refer to these highly compact but still intelligent radio systems as "MicroRadio". The size, performance, physical character, complete software control via generic PC platforms, and commercial electronics base of this technology provided an ideal "tool kit" from which to begin development of the desired Air Force system.

The technical philosophy of the program may be stated as one based on rigorous system design making maximum use of commercial wireless system design methods, modified and extended as needed to suit the Air Force mission. Modern wireless systems have enjoyed outstanding technical success generated by literally thousands of staff years of research and development funded by the explosive financial growth of commercial wireless. The results are not only widely available, low cost, high performance electronic components and integrated subsystems, but a general set of highly effective system design methods that may be cost effectively extended to the special requirements of telemetry. Since a great many of the technical challenges of telemetry system design are very similar to commercial wireless, it makes sound engineering and business sense to use the well proven foundation of commercial wireless system design to the greatest degree possible. As a specific example of the similarity of the problem set, note that one of the strongest technical requirements of commercial wireless is spectral efficiency. Network operators are given a limited bandwidth to work with, and the technology that cost effectively provides the most conversations per MHz per square kilometer is generally the one that stays open for business. With the loss of spectrum to telemetry users imposed in recent years by legislative action and spectrum auction, the same basic requirement to get the most use out of each MHz has been imposed on telemetry. The issue is important enough that loss of spectrum and how to deal with it was the theme of last year's International Telemetry Conference Blue Ribbon Panel.

The software controlled operational concept for the ASMT system is shown in **Figure 1**. The operator(s) may monitor and control the test, including reprogramming data acquisition parameters, from the keypad. The system shall consist of the major components shown in block diagram form in **Figure 2**. The ground station shall utilize PC controlled Base Stations that communicate in upper S band (2310 to 2390 MHz) and in the 2400-2483.5 MHz Industrial, Scientific, and Medical (ISM) band with the airborne station. The airborne station consists of an Integrated Telemetry Unit (ITU) with an intelligent 2300-2500 MHz transceiver, data processing board for encryption and storage, and one or two 18 channel data acquisition boards. There are built in generic sensors for acceleration, air pressure, and temperature, and the data acquisition boards interface over cables to off board sensors mounted where needed by a particular test. There are also optional highly compact Wireless Sensors that may be mounted up to 100-200 feet from the ITU in order to eliminate the need to run long wires, or that can be used on released stores for separation studies. The Wireless Sensors consist of a size reduced 902-928 MHz ISM band transceiver and a six channel data acquisition board. A wireless sensor capable ITU trades off one of its two data acquisition boards for a 900 MHz transceiver to serve the wireless sensor to ITU link. By using 900 MHz for the wireless sensor to ITU link and 2.3-2.5 GHz for the air to ground link, both links may be simultaneously operational and throughput is maximized.

The air interface chosen is one based on Time Division Multiple Access (TDMA) using Gaussian Minimum Shift Keyed (GMSK) binary frequency modulation. The GMSK modulation achieves 1.3 bits

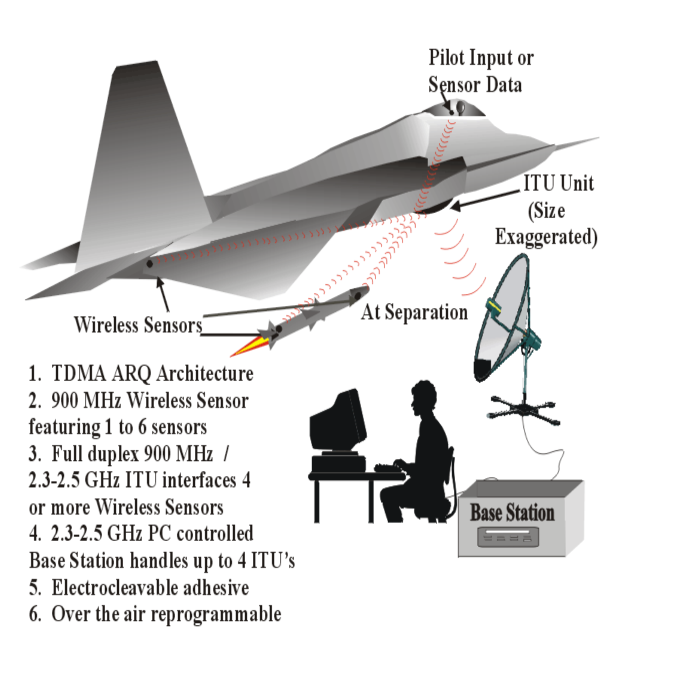

Figure 1: Advanced SubMiniature Telemetry System concept of operational use.

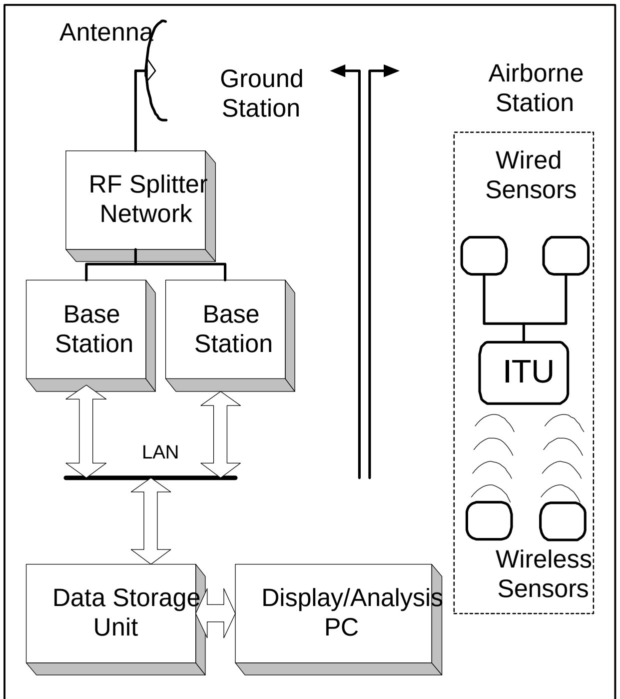

Figure 2: Advanced SubMiniature Telemetry System block diagram.

per second per Hz with BT=0.3. The system covers the full upper S band from 2310 to 2390 MHz in narrowband form with 150 kHz channels, and covers the 2400-2483.5 MHz ISM band in frequency hopping spread spectrum mode. The maximum airborne (mobile) transmit power is 1 watt, and the maximum planned ground (Base Station) power is 3-4 watts. The relatively low transmit powers used in combination with continuous transmit power level control minimizes interference to other users. The modest data rate per ITU of 195 KBPS, narrow receive bandwidth of approximately 150 kHz, and high gain ground antennas allow ranges up to 50 miles at these power levels. Automatic retransmission of any data packet errors ensures high data integrity without having to go to excess in transmit power, antenna gain, Forward Error Correction, or other link parameters.

Typical measurement requirements include acceleration, strain, air pressure, and temperature. Because established sensors are expensive within the context of this program, an important element of the program is exploring the use of low cost commercial sensors. An example is the use of ultra low cost automobile air bag accelerometers, individually calibrated in production with calibration data stored in EEPROM. Such accelerometers cost less than $20 per axis, as compared to approximately $1500 for a typical aerospace quality 3 axis accelerometer. New generation mil-spec MEM's and integrated sensors are also becoming available for prices that are much less than established mil-spec components, and are being studied under this program. Data acquisition bandwidths that read the sensors are typically from tens of Hz to a few kHz. Since any telemetry system is a data pipe with a fixed upper limit on throughput, a data acquisition system that allows flexible trading of sample rate and resolution is the ideal solution to getting the most operational use out of the link. The sigma delta analog to digital conversion technique provides this trade between sample rate and resolution in a convenient "system on a chip" form. These intelligent chips have serial interfaces that allow reprogramming of sample rate and resolution at any time, including over the air reprogramming during the mission.

The equipment design emphasizes compact size and low cost to an extreme degree. Current plans are for Wireless Sensors with data acquisition, radio, controller, and battery to be 1.3 X 2.5 X 0.7 inches. Integrated Telemetry Units independent of aerodynamic tapering are approximately 3.0 X 3.5 X 0.75 inches (1 inch thick for Wireless Sensor capable ITU's), including rechargeable battery. The Base Stations are PC based units and are approximately the size of a compact desktop PC. Costs are in line with commercial electronic products of similar complexity.

System software is based on commercial C (for firmware) and C++ Windows compatible (for PC's) compilers. The airborne end of the link (mobile terminals) uses generic 16 bit one time programmable microcontrollers. The ground end (Base Station) uses similar microcontrollers on each of the four transceivers in each Base Station as well as a Pentium controller board that serves as the Base Station master controller. The Base Stations are networked to operator interface and data storage PC's over a generic local area network such as Ethernet. Commercial database software such as Microsoft SQL-Server provides efficient, cost effective relational database management. Custom software converts the databased information to IRIG-106 format when desired by the user for compatibility with existing analysis methods and software.

WIRELESS SYSTEM DESIGN

The search for greater spectral efficiency in recent telemetry research has focused on more advanced modulation formats such as GMSK, QPSK, and FQPSK. It is of course true that these forms simply use less spectrum, taking standard FSK from its basically 1 bit per second per Hz occupancy to about 1.3 bits per second per Hz for BT=0.3 GMSK to about 2 bits per second per Hz for the various forms of QPSK. However, we submit that the advances cannot stop there, and that from a wireless system design perspective the first and most important change needed from standard telemetry practice is to provide transceive capability at both ends of the link to allow **intelligent link management**. It is basically agile control of the link to provide transmit power level control (transmit only the power needed on a second by second basis to minimize interference) and frequency agility (dodging interference) that allows commercial wireless systems to attain their high density of users. We also propose the use of multiple lower power and data rate and thus lower channel bandwidth links instead of a single faster link to fit these links into smaller gaps of available spectrum. However, we plan one major departure from standard wireless practice, which is the use of packet radio style retransmission (Automatic Repeat reQuest, ARQ) of bad data blocks in conjunction with Forward Error Correction (FEC). Though such bad blocks are unavoidable in a fading environment, retransmission is not used in digital voice cellular because of the latency. Wireless data acquisition can typically afford a small fraction of a second latency. We have adopted Time Division Multiple Access (TDMA) because in practice it attains the highest spectral efficiencies attainable in cost effective year 2000 technology, it provides a convenient frame structure supporting the use of ARQ, and it is highly scaleable in support of a large range of data acquisition sample rates. An argument can be made in favor of Code Division Multiple Access (CDMA) based on direct sequence spread spectrum, but it is more expensive, difficult, and power consumptive to implement.

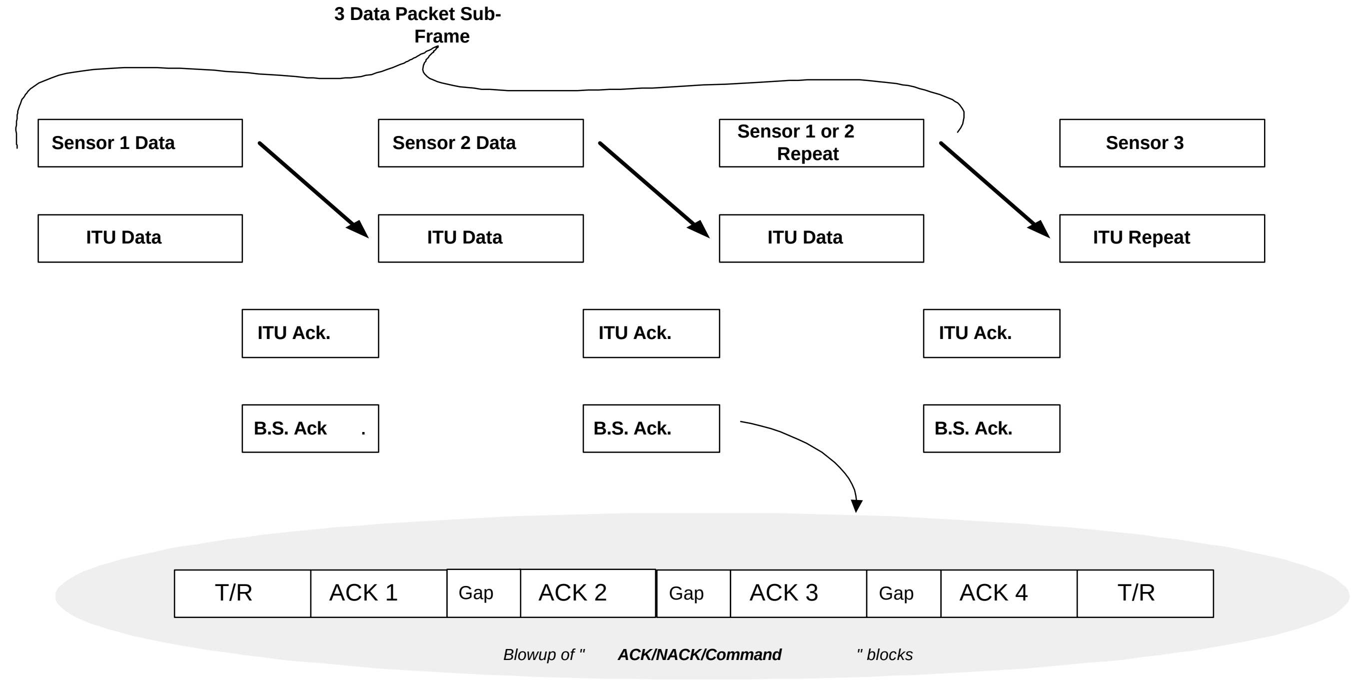

Figure 3: TDMA frame structure showing built in acknowledgement and retransmit from both Wireless Sensor to Integrated Telemetry Unit (900 MHz) and Integrated Telemetry Unit to ground (2.3-2.5 GHz).

TDMA Frame Structure. The planned frame structure is shown in **Figure 3**, with packets shown as they would be spaced in time, with simultaneous packets aligned vertically. After each packet of any kind, the receiving end acknowledges correct receipt, or not, with an "ack/nack/command". Incorrectly received packets are stored for retransmit during the assigned retransmit interval, here shown as one retransmit packet per two standard packets per subframe. Each ITU has 18-36 wired sensor channels (1-2 data acquisition boards), or 18 wired sensors in the case of a Wireless Sensor capable ITU that can also access multiple Wireless Sensors. Packets are approximately 11.6 ms long and each 3 packet "subframe" with overhead is approximately 44 ms long. The subframe is adopted because a mobile (Wireless Sensor or ITU) does not have to use a slot on each subframe, and thus larger numbers of mobiles can be multiplexed in if their data rates allow. A "superframe" is formed by the period of slowest repeat of any mobile terminal in the system, such as taking one packet per ten subframes. The packets contain 283 bytes of data, of which 220 bytes are actual payload. In data flow terms each packet provides slightly over 40 KBPS after Forward Error Correction, retransmit, and other overhead, so each ITU and Base Station transceiver can process about 80 KBPS, and each toaster sized Base Station can process about 320 KBPS while maintaining full ARQ and high data integrity. Multiple Base Stations are used when a test requires more total data flow. The TDMA frame structure and complete software control allows the data rate to broken up and assigned to individual ITU's and sensors in an extremely flexible manner using narrow (150 kHz) channels that make highly efficient use of available spectrum. For example, each Base Station might normally interface to four ITU's, but if the sensed data were sampled at a lower sample rate or resolution, then each Base Station might take eight or 16 ITU's. The system software will guide the operator in making these assignments depending on the sample rates and resolutions requested by the test engineers.

Statistical Retransmit Analysis. The use of ARQ to maintain high data integrity for file transmissions has been standard art for decades. When transmitting files the goal is generally perfect transfer, taking whatever time is required. For data acquisition via telemetry, with a constant input rate of acquired data, it will not prove possible to achieve perfect transfer due to the fact that there must be some limit on the number of retransmit tries allowed. Let us define the packet error rate before retransmission as the "raw packet error rate" and the residual error rate after retransmission as the "corrected error rate". Statistical link analysis can predict and allow minimization of the raw packet error rate, depending on such factors as statistical fade model, antenna gain and angle above ground (aircraft altitude and distance), transmit power, data rate, and other RF parameters. Analysis is ongoing to develop extensions of standard wireless link analysis methods to accurately predict raw packet error rate for the air to ground telemetry environment. Once the raw packet error rate is optimized, it will still generally prove to be undesirably high, particularly for lower power links such as proposed here for battery powered "Peel and Stick" usage and interference avoidance. However, it is possible to attain huge reductions in error rate via a "part time" retransmit capability whereby a fraction of frame time is set aside for retransmission. For example, a relatively high and almost unusable raw packet error rate of 10% due to a hostile link can be reduced to a highly usable 1E-6 corrected packet error rate in our planned TDMA system. If raw packet error rate can be held to a few percent, corrected error rate can asymptotically approach zero.

Having failed to find an analysis of these effects in the public literature, we outline below the form of our own analysis. We find two major contributions to corrected packet error rate, which we define as the "Burst Lost Packet Rate" L_{prb} and the "Timed Out Lost Packet Rate" L_{prt}. The burst lost packet rate is the more obvious one, and refers to the situation where memory storage is limited and old bad packets that have failed to successfully retransmitted are eventually discarded to make room for newer errors. We begin by assuming a TDMA frame with some number of slots assigned for "standard" (first time) transmissions and others for retransmit transmissions, such as two standard and one retransmit slot per frame. The statistics of successful reception of a particular packet constitutes a Bernoulli trial (see ref. 4 or any senior or graduate statistics text) by definition, with the experiment being successful reception one time out of multiple trials. A set of frames thus shows statistics that follow a Binomial distribution, just as a set of "F "dice throws constitutes a series of Bernoulli trials that shows a similar distribution. So, if there are F transmissions (standard or retransmit), and each has probability P_f of failing and P_s = 1 - P_f probability of succeeding, then the odds of "i" successes out of "F" trials is given by:

$$P(i\_successes\_of\_F\_trials) = \frac{F!}{(F-i)!\,i!} Ps^{i}(1-Ps)^{F-i} \quad (1)$$

Similarly, the odds of k failures is given by:

$$P(k\_failures\_of\_F\_trials) = \frac{F!}{(F-k)!\,k!} Pf^{k}(1-Pf)^{F-k} \quad (2)$$

Now, to get buffer overflow we must get a burst of errors over a period of time of interest. If we assign a finite number "M" as the maximum packets a mobile can buffer for retransmit purposes, then the fraction of the total error rate that is due to single overflows (the buffer only went over its limit by one packet) over F frames is:

BurstLostPacketRate1= (Prob(k=M+1 of F transmits)*Prob(zero successful retrans of FX/Z attempts) + Prob(k=M+2)*Prob(one successful retransmit) +...+Prob(k=M+X of F transmits) + Prob(X-1 successful retransmits))*1/F (3)

A similar result may be written for the contribution from two overflows, three overflows, etc, up to a maximum possible of F-M overflows. Then, each of these equations may be written in terms of the product of two Bernoulli trials, where one is for standard failures and the other is for retransmit successes. For the general case of "N" overflows we will get (after a few pages of algebra):

BurstLostPacketRateN=

$$\frac{1}{F}\sum_{i=N}^{F-M} Binom(F, M+i) Pf^{M+i}(1-Pf)^{F-M-1} Binom(\frac{FX}{Z}, i-N)(1-Pf)^{i-N} Pf^{\frac{FX}{Z}-i+N} \quad (4)$$

where Binom(A,B) = A!/(A-B)!B!.

We may then note that the total BurstLostPacketRate is given by eq. (5) below which may be coded in software for numerical evaluation.

BurstLpr = (1/F)(BurstLPR1 +2*BurstLPR2+...+NBurstLPRN+...+
(F-M)BurstLpr(F-M)) (5)

The other error source following retransmission, the Timed Out Lost Packet Rate L_{prt}, will usually dominate in systems with plenty of memory (can store more than 6-10 bad packets in the mobile for retransmission) and is more easily derived. Intuitively, it is the error rate that results from packets that are stored in memory, but which the system never gets to retransmit because of limited retransmit opportunity, additional incoming errors, and some time limit on storage based on the practical limit of distinct numbering of packets. The derivation is as follows. Let there be B_p bad packets (standard packet failures) from a mobile terminal of interest over an interval of time containing F frames, noting that if the raw packet failure rate is P_f, then the expected number of bad packets is $B_p = F*P_f$. The number of retransmit opportunities per mobile terminal sharing the frame over the period of time where the F frames occur is fairly enforced by rule to be FX/Z, where X is the number of retransmits and Z the number of standard packets (and mobiles) per frame. Then over a large number of frames F the number of retransmit opportunities N_{rt} per failed standard transmit is at least (other mobile retransmit slots may be "borrowed" if not needed) the number of total retransmit opportunities per mobile divided by the failed standard packets B_p per mobile, or:

$$N_{rt} = \frac{\frac{XF}{Z}}{B_p} = \frac{\frac{XF}{Z}}{P_f F} = \frac{X}{ZP_f} \quad (6)$$

Given this number of retransmits, the probability of a packet still being failed after all these retransmits is P_f^N_{rt}. Thus the minimum probability of a successful retransmit P_{srt} of a given error is:

$$P_{srt} = 1 - P_f^{\,N_{rt}} \quad (7)$$

Now, over a period of time so large that any errored packets that have not made it through the link yet are considered to be errors, the Lost Packets are given by $L_p = B_p - C_p$ (eq. 8) where C_p are the corrected packets over that interval. But $C_p = B_p P_{srt}$ (eq. 9) and the Timed Out Lost Packet Rate in lost packets per transmitted packet (not lost packets per second) is given by $L_{prt} = (L_p/F)$ (eq.10). Now substituting (6), (7), (8) and (9) into (10) yields:

$$L_{prt} = P_f^{\frac{X}{ZP_f}+1} \quad (11)$$

L_{prt} is a fractional lost packet rate, so to get the lost packet rate per second L_{prt} is multiplied by the subframe rate.

In reviewing numerous ARQ academic papers the result of eq. (11) has not been encountered. However, none of the existing ARQ systems reviewed have a continuous throughput requirement (they take as long necessary to send a perfect file), nor do they use the shared retransmit concept. So, being able to take very long times, having low raw error rates, and having a large effective "X/Z" ratio takes these systems to the point where this error rate is essentially negligible. For example, even in our frame structure of limited retransmit of 1 retransmit packet per two standard packets, a raw packet error rate of 2% would yield L_{prt} = 6.7E-45, a completely negligible number. But let raw error rate increase to 20% and L_{prt} = 3.58E-3, a big improvement over 20% but probably an unacceptable corrected error rate for most applications. Thus attaining a reasonably low raw packet error rate is still highly desirable. This is investigated in the next section, with a graph of corrected error rate shown in **Figure 4**.

Statistical Link Analysis and Packet Error Rates. The radio channel in a mobile communications system presents quite a hostile environment to telemetered data, primarily due to signal blocking and the multipath fade problem caused by destructive cancellation of reflected signal components (see ref. 1 chap. 4 and ref. 2 pages 12-18 for good discussions). This problem has been studied intensely for mobile communications in general, and effective methods for minimizing and predicting errors has been developed for the most common applications (unfortunately, this does not include air to ground telemetry). Due to the non-deterministic time varying nature of the received signal under multipath conditions, the received signal strength must be described using probabilistic methods. The most commonly used probability density function employed is the Rayleigh (ref. 1, p.172), which results when there is no direct wave component and communications is actually carried out with nothing but reflected waves. Of course, in mobile communications there is often a direct wave component, but since most systems must remain operational in the worst case of reflected wave communications only (a cellular handset cannot always be controlled to have a direct line of sight to the Base Station), the Rayleigh is commonly used for system design. However, this worst case assumption is often too conservative for telemetry systems, particularly the air to ground case with highly directional antennas at the ground end of the link. With both a line of sight path and directional attenuation of reflected components this link is friendlier than typical mobile links such as cellular, except for the faster fade process due to higher velocity. But "friendly" is a relative term, there are still many fades in a typical air to ground link, and definite steps must be taken in system design to deal with them. In the 1999 Proceedings of the International Telemetry Conference there were three excellent papers dealing with the modeling of the air to ground telemetry link, listed here as references 7, 8, and 9. However, as good a start as these papers are in starting the process of bringing analytic modeling to the air to ground telemetry link, considerably more work is needed to complete the process and provide the tools needed by the system designer to optimize system performance. In particular, we note that the situation of a dominant ray and several interfering multipath terms has an envelope fade described by the "Ricean" probability distribution. The parameters needed to complete a Ricean model of the air to ground link are not known by the authors to be available, and this appears to be a fertile area for future research.

There are numerous statistical quantities needed to model the radio link failure statistics (ref. 1 chap 4), but here there is only space to review a few of the most important. Some of the more critical are the probability density function (PDF) of the received envelope, the average duration of a fade (ADF) and the level crossing rate (LCR). The PDF describes the time varying nature of the received envelope in statistical terms, and its integral gives another commonly used type of statistical function, the cumulative probability distribution (CPD) (ref. 4, p.36), as shown in (12) below where p(r) is the PDF of random variable "r". The CPD evaluated at a particular value "R" gives the probability that the envelope amplitude "r" at any point in time is less than R.

$$CPD = P(r \le R) = \int_{-\infty}^{R} p(r)dr \quad (12)$$

The ADF is the average period of time that the received signal envelope is below a level R. The LCR is the expected rate at which at which the envelope of the received signal crosses a specified level R, in a positive going direction. Knowing the ADF and LCR give an intuitive idea of the odds of any given data packet suffering a fade that would overcome Forward Error Correction (FEC) and destroy the packet, necessitating a retransmit. For example, for the case where LCR is much slower than the packet time, it would be rare for more than a single fade to occur during a packet, and if the FEC can correct significantly more than the bits lost during an average fade then the number of lost packets would be greatly reduced. For higher velocities and longer packets times where LCR exceeds the packet rate, then it is intuitively clear that the fraction of the time the signal is below threshold is (ADF)(LCR), and setting the condition where the fraction of bits that FEC can correct is F_{FEC}>(ADF)(LCR) would again result in the majority of packets surviving the fade process. Furthermore it may be shown (ref. 2 p. 35) that (ADF)(LCR) = CPD, thus neatly capturing many of the statistics of the fade process.

Having a full statistical description of duration of fade and level crossing would allow a detailed understanding of packet failure statistics, and a resulting optimization of system design. We are attempting to develop such modeling internally (it may be available elsewhere, but to our knowledge is not developed into applicable design technique form), and can show some preliminary results. Assuming each received signal component of the set that added to make the envelope are normally distributed and the power spectrum of received multipath waves is symmetrical around the carrier frequency, the received envelope can be determined following the steps in references 12-15, leading to the PDF shown below that is commonly referred to as "Ricean" or "Rician".

$$p(r) = \left(\frac{r}{b_0}\right) * I_0\left(\frac{r * E_0}{b_0}\right) * \exp\left\{-\left(\frac{r^2 + E_0^2}{2 * b_o}\right)\right\} \quad (13)$$

Now letting f_m be the Doppler shift and E_o the amplitude of the direct wave component,

$b_n = 2\pi * b_0 * f_m * \left(\frac{1*3*5\cdots(n-1)}{2*4*6\cdots n} \right)$ (14) are the general correlations between received components and

$b_0 = \frac{3*E_0^2}{4}$ (15) is the variance of the envelope strength. Due to the limited amount of space, refer to Jakes (ref. 5) and Rice reference 12, section 2.9, for more detailed explanations of these terms.

Eq. 13 states as long as the received envelope is primarily determined by a single component, a direct wave, the received signal strength has this Ricean PDF. In this scenario, the received envelope rarely experiences very deep fades. As the amplitude of the direct wave decrease, the Ricean PDF degrades into the Rayleigh PDF, defined as:

$p(r) = \left(\frac{r}{b_0} \right) * \exp\left\{ -\left(\frac{r^2}{2*b_o} \right) \right\}$ (16) In this situation, the received envelope is determined by the vector summation of the received signals. This results in deep fades in the envelope due to constructive and destructive interference of the received signals.

To determine the LCR and the ADF, similar mathematical procedures to those used by Rice (ref. 14) to develop these parameters for the Rayleigh fading case are used to pursue the Ricean fading case. We begin with the LCR expressed mathematically as

$$LCR = N_r = \int_0^\infty \dot{r} * p(R, \dot{r}) d\dot{r} \quad (17)$$

where the dot indicates the time derivative and $p(R, \dot{r})$ is the joint probability density function. Carrying this operation out for the Ricean density function in (13) results in (18) below.

$$LCR = N_R = \sqrt{\frac{b_2}{2*\pi}} * \left(\frac{R}{b_0} \right) * I_0 \left(\frac{R * E_0}{b_0} \right) \exp \left\{ -\left(\frac{R^2 + E_0{}^2}{2 * b_0} \right) \right\} \quad (18)$$

The ADF is expressed mathematically as (ref. 2 p.35):

$$ADF = \bar{\tau} = \frac{\left(\frac{\int_{-\infty}^{R} p(r) dr}{T} \right)}{\left(\frac{N_R}{T} \right)} = \frac{\int_{-\infty}^{R} p(r) dr}{N_R} = \frac{CPD}{LCR} \quad (19)$$

Note in (19) that a complete grasp of ADF and LCR implies a complete grasp of CPD and thus PDF also, thus granting a high degree of understanding of the full statistics of the fade process. Substituting equations (13) and (18) into (19) results in:

$$ADF = \bar{\tau} = \frac{\int_0^R \left(\frac{r}{b_0}\right) * I_0\left(\frac{r * E_0}{b_0}\right) * \exp\left\{-\left(\frac{r^2 + E_0^{\ 2}}{2 * b_0}\right)\right\} dr}{\sqrt{\frac{b_2}{2 * \pi}} * \left(\frac{R}{b_0}\right) * I_0\left(\frac{RE_0}{b_0}\right) \exp\left\{-\left(\frac{R^2 + E_0^{\ 2}}{2 * b_0}\right)\right\}} \quad (20)$$

More work is needed to validate and extend these models into usable design methods, and to determine the full parameters experimental data must be collected. We hope to be back next year with more extensive Ricean modeling results, including design methods that allow for precise use of the fade statistics to optimize system design. Recall from the retransmit analysis that there is a huge drop in corrected error rate as raw error rate drops below about 20%, so control of raw error rate and by implication understanding of the fade statistics really pays off.

In the meantime we conclude with a numerical example of raw and corrected packet error results based on the worst case Rayleigh model. It can be shown (ref. 6) that the probability of link failure for a set of bits (message block or packet) is quite binary, such that over a fade window of a few decibels the link goes from basically "good" (error correction capability of a few percent can handle the errors) to "bad" (error correction is overwhelmed). Then the raw probability of packet failure for a packet of time length "L" with transmitter velocity V, carrier wavelength lamda, and threshold to average received power "r", can be shown to be (ref. 6):

$P_f = 1 - k_1 k_2^{\ VL}$ (21), where

$k_1 = e^{-0.693\, r^2}$ (22), and

$k2 = e^{-(\frac{2r}{\lambda})k1}$ (23).

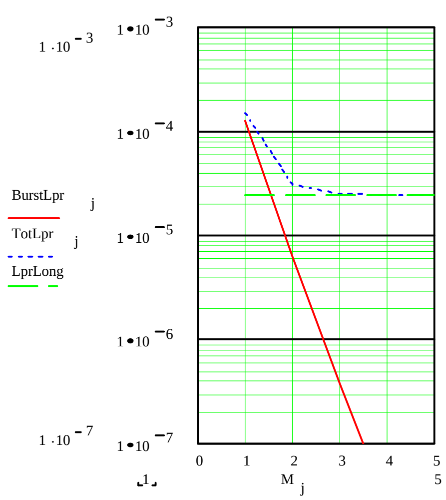

Figure 4: Example ARQ corrected packet error rate as function of memory buffer size.

Table 1: Raw packet error rate vs. velocity , Rayleigh faded 50 dB fade margin.

Vel (MPH)	Pe
200	0.0635
400	0.123
600	0.179
800	0.230
1000	0.280

To estimate the impact on raw packet error rates for Rayleigh fading while attempting to not be too drastically worse than Ricean, let us make the illustrative assumption that air to ground Ricean fading at a given threshold level is 20 dB improved over Rayleigh as far as packet failures are concerned. For Ricean to be this much better than Rayleigh is not optimistic, as reference to ref. 2, p. 35, Fig. 1-13 will show that for a dominant direct path it is usually highly pessimistic. It is also overly conservative at higher speeds (predicted error rate is higher than actual) because some high speed fades will be so fast that the FEC will not be overcome and the packet survives. Link analysis shows that with expected receiver performance at 200 KBPS, 1 watt ITU power, and with 38 dB gain ground antennas, that a range of 47 miles may be obtained with 30 dB fade margin. When 50 dB fade margin is used in eq. (21) above (approximated 20 dB estimated difference), the raw (before ARQ) packet error rates given in **Table 1** result for our TDMA packet parameters. This error rate includes failures due to both the packet and its acknowledgement. **Figure 4** depicts the Burst Lost Packet rate (BurstLpr), the Timed Out Lost Packet Rate, and the sum of the two (TotLpr), versus the number of bad packets "M" for which memory is available to store, for the case described above at 400 MPH with raw packet error rate of 12.3%. At this fairly high Rayleigh packet error rate we find the time out error rate tends to dominate (the flat line at 2.5E-5). But even so the improvement in error rate is over a factor of 4000**,** and for mild reductions in raw error rates the corrected error rate will decrease extremely rapidly.

RF CIRCUIT DESIGN

The physical design requirements of the program include not only unusually compact size and low power consumption, but also the need to operate across an extended temperature range using commercial components. A number of innovations have been developed to enable this implementation, details of which we hope to present in a dedicated article on RF circuit design advances in the ASMT system at the 2001 ITC. For now we note that among the special features of this system are GMSK modulation implemented very efficiently within the synthesizer phase locked loop, new methods of highly efficient power amplifier design, and specialized techniques to extend commercial components to the required temperature range. The receiver design methods are generally classical, being double conversion superheterodyne with non-coherent demodulation.

DATA ACQUISITION

The always limited bandwidth and transmit power of a wireless communication system constrains the maximum data throughput, and this constraint often proves to be a limiting factor on raw technical performance of a system. This bandwidth constraint often dictates compromises in system performance, requiring tradeoffs among the three data acquisition parameters which directly drive the data throughput rate: resolution, sample rate, and channel count. For a given bandwidth, as resolution increases, sample rate and/or channel count must decrease to sustain the same (limiting) data rate. A system can be designed to yield specific performance with respect to these three performance parameters. However, when a system is designed with a fixed selection of data acquisition parameters it becomes dedicated to a specific application and often not readily adaptable to other applications. The ability to alter the balance between the three data acquisition parameters would improve flexibility of a given system, and even better would be the ability to alter these parameters "on the fly", to adapt the system to specific conditions encountered during a test.

A high-resolution ADC can be implemented by several different practical methods. Integrating ADCs have traditionally been used where very high resolution is needed and can be gained at the expense of speed. Such ADCs typically offer speeds of only a few dozen samples per second, but offer resolutions to as much as 26 bits. For faster conversions, the successive approximation ADC has been king, offering speeds well into the megasample-per-second range, but requiring expensive tight-tolerance processes to achieve accuracy, and high power consumption to achieve the higher sampling rates. Any given ADC of either of these types typically offers a fixed limit on both resolution and sampling rate. However, the sigma-delta conversion method has been used in the past few years to implement a number of new ADCs which offer much higher speeds than integrating converters, combined with equivalent or higher resolution compared to successive approximation converters, all for lower cost and power consumption than either of the earlier types. The sigma-delta conversion process is performed by a modulator combined with a digital filter. The modulator produces a high-speed stream of digital values of ones and zeroes, attempting to match the pulse density (average voltage of the output pulses) to the incoming analog signal. The analog portion of this process requires little more than an analog comparator, which is readily implemented with sufficient precision in economical CMOS processes. Knowing that the digital pulse train has a density (proportion of one-bits to total bits) which is precisely proportional to the input voltage, the converter trades the very high speed of this digital bit stream for a much higher resolution by applying a digital decimation filter to the pulse train. That is, the digital pulse train might have a sampling rate of 300 KSPS (thousand samples per second) with the relatively poor resolution of one bit. The digital filter converts this pulse train to a much slower (1 KSPS) but much higher resolution (16 bits) data stream, or perhaps even a 200 SPS 24 bit data stream, which corresponds then to the input waveform. Combined with a preamplifier, noise floors below 1 uVrms can be attained.

These new sigma delta converters offer an unprecedented degree of flexibility in their modern "system on a chip" form that takes maximum advantage of their chip DSP, allowing a software controlled tradeoff between sampling rate and resolution. But this flexibility carries much further into the overall system than this simple tradeoff would indicate. The sigma-delta method provides an inherent antialias filter which can often replace specialized antialias filters in a real data acquisition system. In conventional data acquisition systems, the analog signal is normally filtered by one means or another to enforce an analog signal bandwidth between 1/10 and 1/4 Hz per SPS. If the sampling rate is changed, the antialias filter cutoff frequency must be changed in proportion; such programmability can be provided by a number of means such as switched-capacitor filters, analog filters with component switching, etc. With a sigma-delta converter, however, the input signal is actually sampled at a very high rate; and the external antialias filter need only exclude analog signal components above about 1/4 of the modulator sampling rate, or about 75 kHz for a 300 KSPS modulator. The digital filter action provides the antialias filter with respect to the output sample rate. As a bonus, the digital filter's cutoff frequency can be made to automatically change with the output sample rate, typically maintaining a ratio of approximately 4 SPS per Hz. Since the sigma-delta ADC sampling rate is typically set electronically, by programming from a microcontroller, the sample rate / resolution tradeoff can be made remotely.

Since the sigma-delta ADC can be readily implemented in a CMOS process due to its relatively tolerant accuracy requirements, power consumption can be made mainly dependent on the processing speed. Thus a given ADC or ADC section can effectively be turned off to save power simply by disabling its clock. Most currently available sigma-delta ADCs include such provisions, either by pin-programming or via serial programming of the device. Thus control of the number of channels becomes practical, since unused channels can be readily turned off to save power, always a precious commodity in a mobile unit. In addition, simple changes to parameters of the digital filters can provide a programmable gain capability. Reprogrammability also allows for automated adjustments, since more computing power is typically available at a fixed location, and the data stream can then be analyzed in real time and requests relayed via the telemetry control channel to change the data acquisition configuration. For example, where a test sequence includes a particular event involving significant mechanical activity, the sampling rate for accelerations and strains could be temporarily increased for just that portion of the test, indeed increased even beyond the capacity of the telemetry link, and then returned to a low speed at the conclusion of the dynamic segment, allowing the telemetry link to catch up with buffered data.

In summary, the programmable capability allows precision control of the trade between resolution and sample rate, and fits extremely nicely with efficient use (sharing between units) of the TDMA radio channel, providing optimum use of the fixed bandwidth available in a telemetry link. The compact size, low power consumption, and low noise floor (<1uVrms) of sigma delta are also extremely well suited to wireless data acquisition. The sigma-delta method of ADC is thus a very natural fit to the application, and has been adopted for the ASMT system.

DATA PROCESSING

A Data Processing board is included in the ITU stack to provide significant memory storage for bad packets and a data processing engine for compression and encryption of data. Though the retransmit analysis showed that the ability to typically buffer only 6-10 packets is needed to get error rates that are extremely low, long term blocking (such as aircraft shading of the link) and extremely high data rate acquired data can benefit strongly from extra storage. For example, during separation events high sample rates are typical, and a "data bucket" to store large amounts of data until the radio link can "catch up" is extremely useful. The military application also calls for encryption, though the future exact specifications for this are not clear, and it is a thorny problem for many military programs using wireless and other portable equipment. Regarding compression, historically most digital communications systems with significant processing power do benefit from compression, and we expect that to be the case here. Reference 11 shows that compression ratios from 10% to 60% are practical with typical telemetry data.

CONCLUSION

The SBIR funded Advanced SubMiniature Telemetry program has provided not only an opportunity to provide the "Peel and Stick" portable telemetry equipment needs of the U.S. Air Force, but also a vehicle to bring state of the wireless system and electronic circuit design to the telemetry field. Among the major features of the system are:

- Two way radio links with intelligent link management based on modern wireless system design methods to both cause minimum interference and avoid other interference
- TDMA ARQ (packet radio) architecture for scaleable data acquisition, high spectral efficiency, and high reliability data. The key factor to recognize here is that rather than attempting the impossible goal of near zero errors with more transmit power, more powerful FEC, higher gain antennas, etc, it's simply smarter design (and better performance) to achieve the goal of low residual errors by sacrificing a fraction of the throughput for retransmit.

• Wired and Wireless Sensors based on COTS technology that are typically order of magnitude less expensive than established sensors

• Sigma Delta data acquisition for high raw performance and convenient trade off of sample rate, resolution, and channel count

• Complete software control, with capability to alter test parameters and reprogram data acquisition and sensor parameters at any time, and to convert database files to IRIG standard formats

• Covers both upper S band and the 2400-2483.5 MHz ISM band to allow the system to fly when upper S is full and to allow commercial resale for purely ISM band use, with resulting lower cost through higher volume for all users

For the 2001 ITC, we shall plan to offer a set of articles for publication that are devoted to specific issues, including Ricean modeling of the air to ground telemetry link, expanded wireless system design, detailed RF, data acquisition, and low cost sensor design, system software design, data compression, encryption, and operational use results.

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