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DESIGN AND IMPLEMENTATION OF A FLEXIBLE, SOFTWARE BASED TDMA SYSTEM

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Abstract

A flexible, processor based, TDMA station has been implemented. This station and its associated variable data rate modem enables users to implement very complex frame structures under software control. Burst rates of 512 Kbit/s - 8192 Kbit/s are possible allowing the transmitted bit energy from each station in the network to be adapted to prevailing conditions. One proposed application of the station is the transmission of mixed stream and packet traffic using a modification of the FODA [5] technique.

Introduction

A flexible, processor based, TDMA station has been developed for use in a number of advanced communications experiments. The station, which consists of a processor based TDMA controller and a digitally-implemented multi-rate modem, allows individual data packets to be transmitted at bit rates in the range 512-8192 kbit/s. Each packet can also be protected by variable rate FEC coding independently of other packets within the burst.

The station has the following major features:

Flexibility Only those functions which would be too slow or too inefficient to implement in software are implemented using hardware. Many different TDMA systems can thus be implemented with no hardware modification.
Adaptability The controller hardware allows the content of the TDMA frame to be changed on a frame-by-frame basis under software control. It also allows variable FEC coding and symbol rates to be applied to specified parts of each burst. In addition, to allow complete flexibility of data formatting, the modem has been designed to accommodate changes of symbol rate within each data burst.
Expandability The controller is based on the industry standard VMEbus architecture so that additional hardware functions can be added easily if required.

The hardware consists of two controller units (one transmit and one receive) and a burst mode modem. Figure 1 is a photograph showing the transmit controller (bottom) and the receive controller (top). The core software is a firmware package which implements a basic TDMA system on this hardware. This firmware enables a user to set up a wide range of frame structures and network types and it can demonstrate and utilise all the basic features of the hardware. Advanced TDMA systems, which fully utilise the flexibility and adaptability of the hardware, can be implemented by writing additional software.

Part I of this paper contains a brief description of the TDMA station in terms of the architecture of the hardware and of the core software. Part II describes a more advanced system which uses additional software in order to allow the TDMA station to operate with both packet and stream data.

Part I - The TDMA station

1.1 Controller architecture

The TDMA controller is implemented using two VMEbus based racks, one for the transmit controller and one for the receive controller. High speed communication between the two controllers is possible using a SCSI bus. The controller is based on a proprietary Motorola 6800 processor card containing 4 Mbyte of RAM. This CPU communicates over the VMEbus with a number of other cards. Figure 2 is a block diagram of the controller hardware.

This hardware allows users to implement a wide range of TDMA systems in software but it does impose some restrictions. These were necessary since, in some cases, the software would be unable to react quickly enough to external events.

- Each burst must start with a carrier and bit-timing recovery sequence (CBTRS). This is actually a restriction of the modem.

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- The frame must include a reference burst (RB) containing the master unique word (MUW). The MUW maintains synchronisation of the receive hardware frame counter. Apart from the MUW the remaining content of the RB is unimportant to the hardware. Only one MUW must appear in each frame. The maximum frame length is 64 ms.
- Each data burst must consist of a control sub-burst (CSB) followed by a number, possibly zero, of data sub-bursts (DSBs). The CSB is decoded by the hardware and provides information necessary for the reception of the following DSBs.
- In the transmit controller data is received from a number of terrestrial interfaces. A transmit serial interface card provides stream interfaces at 64 kbit/s (G.703) and 384 kbit/s (RS449). The CPU card provides an Ethernet interface and other LAN interfaces can be provided by purchasing proprietary VME cards. Data collected from the terrestrial interfaces is formatted in the appropriate bursts by the CPU. The transmit modem interface card then controls data transfer to the modem.

The precise time at which individual bursts are sent to the modulator is determined by the transmit event memory (TEM). This consists of two memory arrays organised in a ping-pong arrangement, each capable of storing up to 512 events. During one frame the software can update one side of the TEM whilst the other side is being used by the hardware. The software originated events are then used by the hardware in the subsequent frame. Each event within the event memory contains an 18 bit time code, which defines the time at which a particular action will take place (in 244 ns ticks from the start of transmit frame), and a 14 bit function code. The transmit events are used, amongst other things, to open and close the burst gate and to set the modulator output level. A special function code can be used to define the end of the transmit frame by providing a reset pulse to the transmit frame counter.

When the transmit burst gate is open, formatted data is sent from the CPU to the transmit modem interface under interrupt control. This data is then convolutionally encoded and scrambled if necessary. The code rates available are all based on a standard 1/2 rate convolutional code but puncturing logic allows additional code rates of 2/3 and 4/5. The hardware appends the necessary tail bits to flush the decoder at the receive side and will, if required, append a short CRC to each transmitted DSB.

In the receive controller 4 bit soft-decision data received from the demodulator is passed to the convolutional decoder where it is decoded if required using the Viterbi algorithm. The receive hardware also provides a measurement of the estimated channel quality for each DSB by monitoring the recovered data soft decisions. The hardware simply observes the occurrence of particular soft decision levels and accumulates these over a sub-burst. This count is then made available to the software. Assuming a Gaussian distribution an approximation to the short-term BER can then be computed. This method is much faster than counting errors in the UW and it provides a channel estimate which can be used in adaptive fade countermeasures algorithms.

The decoded data stream is then routed to the receive modem interface. This interface uses a receive event memory (REM) to allow real-time reception and decoding of the received bursts. The REM is identical in structure to the TEM. The receive function code contains information on, amongst other things, the initial code rate and symbol rate of the bursts within the frame. A special function code can be used to generate a transmit frame pulse which will reset the event counter on the transmit controller. This is used by slave stations to synchronise their transmit frame to that of the master.

Three unique words are used by the hardware. The master unique word (MUW) is reserved for use in the reference burst only. Receipt of the MUW by the hardware resets the receive frame counter and thus marks the start of the receive frame. The second type of unique word is the traffic unique word (TUW) which is used by all bursts other than the reference burst. Finally, the sub-burst unique word (SBUW) is used at the start of each data sub-burst. Each of the three unique words is composed of the same basic programmable pattern and its complement (UW and 'UW'). The SBUW uses the basic UW pattern whilst the MUW and TUW use followed by either UW or 'UW. The receive hardware accommodates two, software selectable, thresholds for unique word detection. In addition to their synchronisation functions the unique words are used by the hardware for data ambiguity resolution and to reset the optional de-scrambler to its start-up vector.

When a data burst (DB) is received the hardware decodes the control words contained within the CSB and uses them to drive the receiver during the reception of subsequent DSBs. This allows each DSB to have its own particular parameters. The hardware decoding of the information in the CSB is required since software would not be fast enough. It does, however, introduce a slight restriction on the formatting of the data burst.

The decoded and descrambled data is then passed to the software under interrupt control along with a number of status words.
which provide information about the received data. The CPU can then route data to the stream outputs implemented by the receive serial interface or to any other output interface available.

Figure 3 shows a typical frame structure of the TDMA station as implemented by the core software. The RB and DB have been described already and their format is partly determined by the hardware. In order to maintain slave synchronization in the network, the frame also contains a number of reserved slots which are used for short acquisition bursts. These convey information about slave timing to the master.

The core software also provides an extensive menu driven interface to users via the controller console. This menu system allows the user to adjust all of the variable parameters of the system. It also provides a powerful monitoring facility by displaying real-time status information on various display screens.

### 1.2 The burst-mode modem

A special feature of the modem is that it is capable of dynamically adjusting its transmission rate within a data burst. This allows the individual DSBs of a DB to have different symbol rates (and, hence, different energies) as required. The symbol rates available are 512, 1024, 2048 and 4096 kbaud using either BPSK or QPSK modulation formats. A bit rate range of 512-8192 kbit/s is thus available to the system. In order to achieve the variable symbol rate requirements of the modem, a block diagram of which is shown in Figure 4, is implemented using digital signal processing (DSP) techniques.

#### 1.2.1 Stored waveform modulator

In an ideal modulator positive and negative impulses representing the two data states of the incoming binary data sequence are passed through a filter with a Nyquist frequency response. Digitally this is equivalent to the convolution:

\[
F(i) = \sum_{-\infty}^{\infty} I(i-j) C(j)
\]

where \(I(n)\) is the \(n\)th data impulse (±1 depending on the data state), \(C(n)\) is the \(n\)th filter coefficient and \(F(n)\) is the \(n\)th output sample. This function is implemented using a Finite Impulse Response (FIR) filter.

As a consequence of Nyquist's sampling theorem the output of the digital modulator has aliases at the sampling frequency which must be removed by analogue anti-alias filters (AAF s) before transmission. Since the above convolution has only one output sample per data symbol it leads to serious AAF implementation problems because the aliases are at a low frequency with respect to the wanted spectrum. This problem is overcome by increasing the input sampling rate by placing intermediate zero value samples in the input stream. In this way it is possible to obtain output sampling rates which are integer multiples of the data rate. Interpolation by a factor of four for instance would shift the aliases to four times their original frequency so that they become much easier to remove by analogue filtering. Symbol rate changes can conveniently be effected by changing the interpolation rate. In this way the output sampling rate, and consequently, the alias frequency, remains constant. The TDMA burst modulator has a fixed sampling clock of 16.384 MHz. Interpolations rates of 4, 8, 16 and 32 thus provide the required transmission rates of 4096, 2048, 1024 and 512 kbaud respectively.

The interpolated FIR can be implemented relatively simply by using a stored waveform modulator structure. The output results for all possible input sequences are pre-computed and stored in a Read Only Memory (ROM). This acts as a large look-up table which is addressed by the past \(n\) symbols of the incoming data sequence and an interpolation count. The resulting output sample is then available for transfer to the DAC.

#### 1.2.2 Demodulator

In the demodulator the received signal, polluted by noise and other effects, is first downconverted to two quadrature baseband signals by an IF converter. Following anti-alias filtering the signals are sampled by two Analogue to Digital Converters (ADCs) and the resulting complex sequence is passed to the digital processor. The complex samples are then filtered by a FIR filter, the response of which is selected depending on the expected burst symbol rate.

Once the received sample sequence has been filtered the original data stream must be extracted. In order to do this the demodulator must make estimates of several parameters from the received, noisy signal. For a phase modulation such as QPSK the parameters of interest are:

- \(\omega\) an unknown carrier frequency translation (due to oscillator inaccuracy in the satellite and the earth stations).
- \(\phi\) an unknown carrier phase shift (modulo 2\pi).
- \(t\) an unknown timing phase shift (modulo T).

The carrier frequency error consists of two components. The downlink frequency error \( \omega_d \) is due to the satellite frequency conversion error and the downconverter in the local receive station. This error can be up to ±40 kHz but it applies to all of the bursts in the frame. The demodulator removes this by an initial AFC sweep at the start of operation and tracks the slow variations thereafter. The uplink frequency error \( \omega_u \) is more of a problem. It is caused by the variations in transmit frequency at the remote stations and is thus different for each burst in the frame. In a typical system \( \omega_u \) can be of the order of ±4 kHz. The value of \( \omega_u \) must be estimated very rapidly at the start of each burst.

In order to allow parameter estimation, and thus data demodulation, before the arrival of the UW, each burst is preceded by a short preamble. The parameters \( \omega_u \) and \( \phi \) are estimated simultaneously at the start of each burst using the unmodulated carrier preamble. Two types of estimator were considered. The
feedback estimator makes an estimate of the error between the wanted phase and e and adjusts the incoming phase to minimize the error. The feedforward estimator makes absolute estimates of e and then passes these forward to a later stage where an appropriate correction is made. The feedback estimator works well under low signal-to-noise conditions and can accommodate large frequency errors, however, in a burst mode system, it does have a major problem called hangup. This phenomenon leads to occasional very long acquisition times when the initial phase error is close to n. This is not acceptable for a burst mode system where reliable acquisition in a limited time is necessary. The feedforward system does not suffer from hangup but, unfortunately, it is not particularly suited to situations where the initial frequency offset is large. In order to overcome these problems this modem uses two digitally-implemented feedback estimators which are initialised to different states just prior to the expected reception of a burst. Since the initial phase of each estimator is different they cannot both suffer from hangup and so one will always acquire within the time available.

The symbol phase, e, is estimated using the reverse present at the end of the preamble, the FQDA controller to different clock phases are used and the phase giving the largest output is used to initialise the clock tracking loop. Once the symbol phase has been estimated the demodulated data is available to the receive controller.

Part II. An application of the presented hardware

An application of the hardware described above is presented in the following. It consists of a user oriented satellite network interconnecting a number of LANs, placed at different sites. At each site, a pretty small earth station of a 2-3 m dish and an HPA of 10-100 W is installed at the user premises and is connected to the satellite controllers through the modem. The employed payload is that of any geostationary satellite, working in the Ku or Ka bands with a global coverage. The inter LAN packetized traffic is concentrated at each site on a satellite router which is connected to the controller and to one or more LANs (figure 5).

The traffic produced by the various applications disseminated over the LANs may be either isochronous (stream) or asynchronous (datagram). Typical stream applications are: telephony (64 Kbit/s or compressed), videotelephony, tele-education and video-conference (e.g. CCITT H.261 at N x 64 kb/s. with N =1-30), while all the classical EDP applications are substantially of datagram type.

The satellite access scheme, running on the controllers, is a TDMA system based on demand assignment of the channel capacity. The system is able to guarantee to the stream applications the requested bandwidth with a limited packet jitter, while doing a best effort to reduce the queuing delay of the datagram packets. Two priority classes are provided for datagram traffic, allowing a lower delay for interactive traffic with respect to bulk traffic. For all the types of traffic the system offers and maintains the data quality required by the applications and specified in the class of service (COS) parameter. Care must be taken in choosing the type and implementation of the LAN to be crossed by the stream data. Indeed, stream applications are jitter sensitive and some of them, such as telephony (CCITT rec. G.114), are also delay sensitive. The LANs must therefore guarantee, within certain tolerances, the instantaneous bandwidth requested by the applications. Contention based LANs, such as Ethernet, are generally unsuitable to carry stream traffic; however, if the peak of the total load is kept below a limited fraction of the network capacity, the introduced jitter can be acceptable [7]. The jitter accumulated during the internetworking system crossing can be removed by the destination application, using an elastic buffer, at the cost of an additional end-to-end delay. Bulk traffic is much less sensitive than stream to jitter and delay, while the delay caused by the satellite network crossing penalizes but still allows interactive applications.

The name of the access scheme is FQDA-IBEA (FIFO Ordered Demand Assignment - Information Bit Energy Adaptive). The first part of the name is the same of a system previously realized at fixed bit and coding rates [5], because the adopted mechanism for capacity request and assignment of datagram is substantially the same. The second part indicates that the energy contained in an information bit can be increased to achieve the application requested data quality under unfavourable transmitting/receiving conditions (signal fading and/or smaller earth station performance). The information bit energy is changed by varying the up-link power, when possible, or gradually varying first the coding rate and then the burst transmission bit rate. Some ideas of the present system were anticipated in [1, 2].

2.1 The access scheme

The satellite controllers, the modem and the earth station are collectively called a station. Centralized control has been chosen because, while operating in fade conditions (high BER), a distributed system would cause frequent losses of synchronization among the stations.

The control station (the first one that comes up) is called a master. It sends a reference burst (RB) for frame synchronization at the beginning of each 20 ms frame. The RB for frame n contains, in addition to other general information, the plan of the transmission time slots for the stations, valid for the frame n+1. The stream slots are allocated one per frame to the requesting stations up to a global amount (the stream boundary). When this boundary is reached, further requests are no more accepted. The stream request (pipe/frame) is the resulting bandwidth needed by all the stream applications served by the station. The bandwidth of each stream allocation is maintained unchanged until a modification (deallocate) request is sent by the station. The remaining channel capacity is
assigned to datagram services according to an algorithm described in [5] which cyclically assigns a fraction of each request to each station. The datagram requests are computed by each station as the sum of two terms: one proportional to the volume of data present at the station in the datagram queue (the backlog), the other proportional to the incoming traffic rate. Denoting the backlog by \( B \) and the incoming traffic rate by \( I \), the datagram request is expressed as \( R = B + h \) where \( h \) is a temporal constant. The simulation results of figure 6 and 7 are obtained loading the channel with Poisson generators of datagram traffic for 10 stations. They show that the best value of \( h \) is 0.4 s.

![Fig. 6- End-to-end delay averaged over 30 s for various values of \( h \)](image)

Datagram and stream requests can be appended to existing data bursts or sent in pre-assigned control slots, whose minimum frequency is guaranteed. Two control slots per frame are foreseen. They are assigned to the first two active stations having no data burst assignment in that frame and follow a round-robin scheduling algorithm.

A unique slot per station for both stream and datagram traffic is allocated in a frame. This choice induces a jitter in the receive sequence of the stream packets, because the size of datagram allocation varies in each frame. In order to reduce this jitter, the transmission of the stream packets received in a frame to the destination router can be optionally delayed up to the end of the receiving frame.

### 2.2 Behaviour in fade conditions

At each station a table is available containing the transmission characteristics. It gives the transmission parameters (bit and coding rates) as function of the BER range requested by the application (class of service) and of the C/No (carrier power to noise density ratio) available at the receiver. When the fade level changes on a link, the sending station updates the transmission parameters of the packets for each application that uses that link. The increase in data redundancy due to the decrease in C/No requires the assignment of wider transmission slots within the frame. The system reacts to a fade change in different ways for stream and for datagram.

For stream traffic the sending station computes the increased size of stream bandwidth and sends a modified stream request to the master. The master rejects the request if the overall requested stream bandwidth exceeds the high stream boundary, but allows the crossing of the normal stream boundary because of the fade.

![Fig. 7 - Channel delay averaged over 10 stations. Poisson datagram traffic. A step of traffic of 20% is applied to one station for 10 s. The choice of the maximum system bit rate of 8 Mbit/s was due to the necessity of limiting the capacity, and consequently the cost, of the stations. The other economic requirement of fully exploiting the transponder capacity thus imposes a multiscatter access to the transponder itself. Each carrier can be shared in TDMA, using FDDA/BEA or similar systems. The various systems may operate autonomously or may belong to a global MF-TDMA system as, for example, one of those presented in [4].](image)

<table>
<thead>
<tr>
<th>Up-Link freq. [GHz] (CH1)</th>
<th>28.072255</th>
<th>Number of carriers</th>
<th>3</th>
</tr>
</thead>
<tbody>
<tr>
<td>Down-link freq. [GHz] (CH3)</td>
<td>19.475</td>
<td>Total IPFD [dBW/m^2]</td>
<td>-102</td>
</tr>
<tr>
<td>E/S EIRP [dBW]</td>
<td>73</td>
<td>Input Back-off [dB]</td>
<td>8</td>
</tr>
<tr>
<td>Satellite G/T [dBK]</td>
<td>14</td>
<td>E/S G/T [dBK]</td>
<td>27.3</td>
</tr>
<tr>
<td>C/T at satellite input [dBW/K]</td>
<td>-143.5</td>
<td>C/No at E/S receiver [dBHz]</td>
<td>82.6</td>
</tr>
<tr>
<td>Intermodulation C/T [dBW/K]</td>
<td>-140</td>
<td>Eb/No at 8 Mbit/s [dB]</td>
<td>13.3</td>
</tr>
<tr>
<td>Total Up-link C/T [dBW/K]</td>
<td>-145</td>
<td>Modern implementation margin [dB]</td>
<td>1.5</td>
</tr>
<tr>
<td>Up-link Eb/No [dB]</td>
<td>14.4</td>
<td>Eb/No in clear sky conditions</td>
<td>12</td>
</tr>
</tbody>
</table>

Table 1. Link budget for the Olympus Ka transponder. Three carriers at 8 Mbit/s access the transponder in FDMA. The 2.5 m E/S is equipped with a 70 W HPA.

In order to limit the intermodulation interference due to multiscatter access, the satellite transponder HPA must be sufficiently backed-off. The IPFD (input power flux density) at the satellite must be kept constant and the transponder gain is set in such a way to operate with the chosen back-off.
Table 1 shows an example of link budget, using Olympus in Ka band and 2.5 m antennas equipped with a tracking feature. Three carriers are considered.

A C/I ratio due to intermodulation interference of -140 dB/K has been assumed. This figure must be confirmed experimentally. In this example the up-link power control range is 13 dB. The value of the Eb/No ratio (12 dB) resulting from Table 1 is assumed as the reference unfaded value. It allows uncoded transmissions at 8 Mbit/s with a BER of $10^{-8}$ for up-link fade conditions ranging from 0 dB (clear sky) to 18 dB.

The up-link power control is realized, on each carrier system, as follows. The power level of the master is assumed to be the reference level for all the stations. Each station is able to estimate the power associated with each of the received bursts using the signal-level indicator implemented by the modem. This is a sort of fast AGC (automatic gain control), whose low-pass filter bandwidth is chosen to be narrow enough to filter out most of the noise and wide enough to track the shortest burst. The characteristic of such an estimator is given in figure 8.

In the following all the power levels are expressed in dB. In order that all of the stations maintain the same IPFD at the satellite, each station compares the levels of the bursts sent by itself with the level of the reference burst sent by the master. The difference is used to modify the output power of the station’s HPA. This is easily realized by modifying the modem’s IF level according to a look-up table which takes into account a small non-linearity in the transmit chain. Within the up-power-control range each station can fully compensate the up-link fade level. The up-link residual attenuation due to out of range operation is compensated with adequate coding and bit rate. The up-power control range of each station depends on the station power margin. An interesting feature of the system is the freedom of sizing the stations according to the traffic intensity, the climatic conditions and the geographic position.

In order to keep its IPFD constant the master needs a reliable estimation of the up-link attenuation. The estimation is accomplished by comparing the level of the bursts transmitted by itself with a reference level measured in clear sky conditions: let $P_{Ru}$ be the up power reference level (clear sky), and $P_{Rd}$ the received signal reference level (clear sky). The total attenuation $At$ is thus

$$ At = (P_{Ru} - P_{Ru}) \cdot (P_{Rd} - P_{Rd}) \quad [\text{dB}] $$

where $P_{Ru}$ and $P_{Rd}$ are the current transmitted and received powers respectively.

At is split into its up-link and down-link components using the long-term frequency scaling formula [10]. The ratio between the up and down attenuation, due to the on board antenna mispointing error, can be assumed to be proportional to the square of the relevant frequencies ratio.

$$ A_{Mu} \cdot A_{Md} = 2 \log_{10} \left( \frac{F_u}{F_d} \right) \quad [\text{dB}] $$

where $A_{Mu}$ and $A_{Md}$ are the up and down-link attenuations respectively.

The assumption of the same behaviour for the rain attenuation (in the Ka band) introduces an error of less than 0.5 dB over a range of 15 dB. This simplifies the problem because the two effects do not need to be distinguished.

An objection can be made concerning the application of the long-term frequency scaling formula to compute instantaneous values (short-term) of the attenuation ratio. In fact, the above ratio depends on rain type and temperature [8].

Experimental results in the Ku band show that the use of the long term frequency scaling formula leads to a maximum error of 2 dB within the 99 percentile [11]. We assume the same figures, until we will be able to get some measurements in the Ka band.

The error made by the master in transmitting the reference power level is reduced by a distributed correction mechanism. Each station sends the master its estimation of the master IPFD at the satellite input, allowing a $\sqrt{N}$ reduction on the standard deviation of the measurement if the contributions can be assumed statistically independent.

At least one station is always available to assume the role of master, when the current master fading conditions increase over a certain threshold.

2.3 Link quality estimation and dissemination

Each sender station must know the C/No value available at the station to which its data is addressed, in order to choose the correct transmission parameters (bit and coding rates). C/No is computed, knowing the bit rate $r$, as

$$ C_r = r \frac{E_b}{N_0} \quad [\text{dB}] $$

The estimation of $E_b/N_0$ (bit energy to noise power density ratio) is made by each station, on each burst arrival, by detecting the percentage of bits whose magnitude is less than a certain threshold (fraction of the average signal amplitude). In figure 9 the probability that the signal magnitude is less than the threshold is given.

![Fig. 9. Probability that the signal magnitude is less than the threshold $t$ and number of bits for an $E_b/N_0$ estimation with a confidence interval of 0.5 dB at 99%, versus $E_b/N_0$.](image-url)

Two different methods, one direct and another indirect, may be followed for C/No information dissemination. By using the direct method, the C/No value is sent directly by each receiving station to
each sending station. This method allows the maximum precision in the parameter estimation but it imposes an overhead due to the volume of control data requested. In fact, each station must frequently send an update of its fade condition to all the stations which are sending data to it. Moreover, if filtering and/or prediction techniques are employed to help estimate the C/No, each receiving station must process data relating to the link it has with sending stations. In some cases this could be a large computing load.

With the indirect method it is supposed that each station broadcasts periodically only the C/No value relating to data it is receiving from the master. Let us call this quantity \( \zeta_m \). The parameter \( \zeta_{s-m} \), i.e. the C/No available at the receiving station when the sending station is transmitting, results to be [6]

\[
\zeta_{s-m} = \zeta_m + C_{s-m} \cdot C_{m-s}
\]

where \( C_{s-m} \) is the carrier power level available at the sending station, when the sending station itself is transmitting and \( C_{m-s} \) is the carrier power level available at the sending station, when the master is transmitting. The term \( C_{s-m} \cdot C_{m-s} \) represents the sending station up-link attenuation, which is outside the up-power control compensation range.

In figure 9 the number of bits to inspect, in order to get an Eb/No with a confidence interval of 0.5 dB at the 95% level, is given for the signal magnitude thresholds of 5/8 and 6/8 respectively. The choice of the threshold depends on the link quality dissemination method employed. In the case of the direct method the range of Eb/No is 7-12 dB, so the best value of the threshold is 5/8. Using the indirect method the quality of the RB must be determined by all the stations, resulting in a wider range. In fact the RB is always sent at 2 Mbits/ to allow a margin for the faded stations and C/No is 6 dB higher. The resulting range of 7-18 dB implies the use of the 6/8 threshold.

Conclusions

A flexible TDMA station has been implemented. With its associated modem it allows very complex and efficient TDMA systems to be designed. Of particular importance is the system's ability to transmit data packets with multiple data and coding rates. This allows the information bit energy of each packet to be tailored to the prevailing conditions at the required grade of service. The TDMA system can be used to implement a complex system based on the FODA architecture. This allows both stream and packet traffic to be accommodated in the same network. In addition stations can operate with reduced power margins since the system adapt to the transmitted bit energy in real time to counteract fade conditions. Such a system will allow the use of smaller earth stations and so will improve the economics of user-located satellite communications networks.

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