Paper

Proposal and Evaluation of Satellite Communication Systems for Distributed Cooperative Work

Nonmember Masanori Suzuki, (Mitsubishi Electric Corporation),
Nonmember Nobuhiro Kataoka(Tokai University)
Nonmember Kazuyoshi Miyoshi (Hiroshima Kokusai Gakuin University),
Member Hisao Koizumi (Tokyo Denki University)

Satellite communication systems suitable for applications in distributed cooperative work are proposed and evaluated. We have been proposing a new method of multimedia data communication, in which voice and video signals are exchanged via allocated SCPC channels, and burst data via code-division multiplexing (CDM). For burst data access, demand assignment and random access are compared. This burst data is transmitted in the same frequency band as voice and video data. The evaluation clarified that it is possible to support a considerable number of concurrent synchronous cooperation while conserving valuable bandwidth resources and alleviating the costs of expensive transponder usage.

Keyword: Distributed Cooperative Work, Wide Area Network, Satellite Communication System, Single Channel Per Carrier(SCPC), Code Division Multiple Access (CDMA)

1. Introduction

Groupware exploiting computer and communication technology holds out the possibility of nullifying differences in time and space, providing support for cooperative tasks by persons extended over broad geographic areas.

Groupware is classified into synchronous types supporting real time one-to-one cooperation and asynchronous non real time cooperative work. In synchronous communication (like video conferencing), a presentation datum (page) must be transmitted as soon as possible by burst data.

Asynchronous communication does not require real time transmission and data will queue in a server.

Usually, long distance traffic will be carried on tight (small margin) lines because of expensive tariffs. Under such circumstances, the burst data may cause chaos in the network especially in the most active business hours.

Generally speaking, satellite communication networks are used for long distance links, and satellite channels can also be used in time of disasters and emergencies as alternate routes to substitute for congested ground channels, and for overseas communications with business offices in Asia and elsewhere under the coverage of the satellite.

The objective of our study is to make clear the possibility of sending the burst data efficiently on satellite communication channels. In recent high-output communication satellites, when signals are received by earth stations with VSAT class antennas 1 to 2 meters in diameter, the satellite transponder operation is normally frequency limited, and there is excess power capacity. By transmitting voice and data signals by Single Channel Per Carrier (SCPC) up to the maximum capacity determined by the frequency limitation of the satellite transponder, and transmitting burst data at the same frequency by Code Division Multiple Access (CDMA) using this excess power, valuable frequency resources can be conserved and the cost of using expensive satellite transponders can be reduced.

In the satellite communication system proposed in this paper, video/audio and low speed data are transmitted via SCPC channels with a connection request. For the burst data transmission including text and image data, both demand assignment (DA) and random access (RA) in code division multiplexing (CDM) schemes are compared. The former allocates vacant channels on request from a transmission-demanding earth station and it is recommended for heavier traffic.

The latter transmits randomly when an earth station senses satellite transponder excess capacities to be available and it is applicable for lighter traffic.

Clearly, frequency-multiplexed signals such as SCPC can be transmitted in the same frequency band as CDMA signals, but there have not to date been any quantitative studies on concurrent transmission making active use of this. In previous studies, the authors announced calculations of capacities for concurrent transmissions of CDMA signals for random access cases using simplified traffic analysis models[1][2][3][4]. In this paper, we conducted systems analyses on satellite communication systems for distributed cooperative work using a detailed traffic model derived by one of authors[6], and we propose a system configuration to save transponder cost and frequency resources. In this work, we attempted to handle more detailed traffic analysis models. In Sec.2, a typical example of our proposed system for business applications is shown that was realized by usual network equipment.

In Sec.3, a traffic model to be used in the following studies is derived from our previously measured data on groupware tools. In Sec.4, CDMA data transmission capacity is calculated with simultaneous transmission of SCPC and CDMA. In Sec.5 and 6, burst data transmission capacity and delay time for DA and RA are calculated within the CDMA data transmission capacity, respectively. In Sec.7, DA and RA are compared. Sec.8 (conclusion) stress the effectiveness of our proposal for distributed cooperative works.

2. System Configuration and Traffic for Distributed Cooperative Work

The proposed multimedia network systems can be realized by ordinary business networks which consist of ground and satellite communication networks as shown in Figure 1.

Satellite communication networks usually have a hub at a central business location, with small earth stations called a VSAT (Very Small Aperture Terminal) system located at outlying sites.

A DAMA (Demand Assignment Multiple Access) unit with either dynamic (exchange) or static channel allocation functions is placed at the hub station.

Within a local area, data terminals which will be used for asynchronous cooperation are connected to the usual LAN, and data are stored in a group (gateway) server which has replicas of remote databases to reduce WAN traffic, and are forwarded when the wide area network is open.

Multimedia terminals, which handle voice and video besides data for synchronous cooperative works, are connected to an ATM LAN and the burst data will flow through the group server where routing to ground or satellite network is selected. Voice and video traffic of the multimedia terminals go to the PBX through an ATM interface(CLAD). Usual telephone sets, which are connected directly to the PBX, are not shown here.

Voice, video and data to other business locations through ground networks are multiplexed/de-multiplexed by a digital multiplexes. This network configuration, consisting of the usual LAN, ATM and PBX is rather conservative, and a more combined and integrated approach is becoming feasible.

3. Estimation of Communications Traffic in Distributed Cooperative Work

We have measured the basic data transmission parameters for typical asynchronous and synchronous groupware for distributed cooperation for years. The asynchronous tool is groupware (Lotus Notes) for data sharing, and the synchronous tool is a PC conference system (Proshare).

Asynchronous and synchronous data transmission efficiencies are measured by "LAN Decoder" which is a traffic measurement tool as shown in Table 1 and 2, respectively. In this Table, information volume is measured by file size in a personal computer.

The amount of traffic that will actually be generated by cooperative work dispersed over a wide area is estimated below.

(1) Telephone

A 3.4 kHz band width telephone quality voice message is transmitted by 32 kb/s ADPCM (Adaptive Differential Pulse Code Modulation).

Table 1 Data transmission parameters with asynchronous tool

Information Volume(kB)	10	20	40	60	100
Transmission Volume(kB)	19.9	31.5	55.1	78.8	125.5
Transmission Efficiency(%)	50	63	73	76	80

Table 2. Data transmission parameters with synchronous tool

	File transfer		White board	Shared applications	
Information vol. (kB)	100	1000	100	0.02	0.2
Transmission vol. (kB)	73.8	677	144	14.8	7.1
Transmission efficiency(%)*	135	148	69	-	-

* Figures over 100% reflect text compression.

With recent voice compression technology, 7 kHz bandwidth AM radio quality can be realized. Bit Error Rate (BER) of less than 10^{-4}, will be sufficient for voice communication.

(2) Video and low speed data

For small windows (some 100 x 100 bits), moving pictures will be sent at 64kb/s that corresponds to 1.5Mb/s in full frame (some 400 x 600 dots) moving pictures where the mobility is sufficient for TV conference applications.

(64 kb/s x 400/100 x 600/100 = 1.5Mb/s)

BER of less than 10^{-6} will be desirable for this application.

(3) Burst Data

The occurrence of data transfer is initiated manually and the probability is considered here as independent of time, following a Poisson distribution. Data length distribution is exponential, which means the probability of end of data transmission is independent of time. The number of services is s. This traffic model is denoted.

Then, typical communication traffic per terminal supplemented by experience is indicated in Table 3.

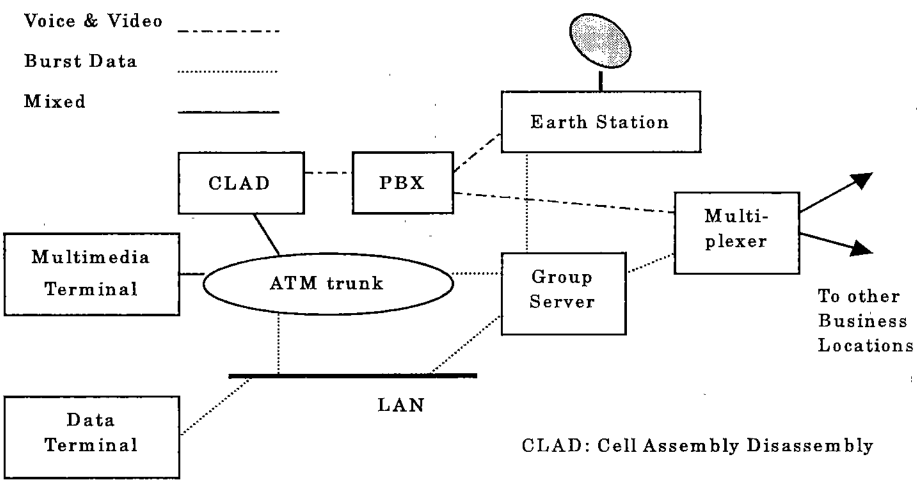


Figure 1 Proposed network configuration as M/M/s by the well-known Kendall notation

Table 3 Typical terminal communication traffic

	type	average data vol./transmit.	occurrence of trans./h
async.	replica	50.0kByte	0 ~ 1
sync.	file	15.5	1 ~ 3
	board	1.0	5 ~10
	shared appli.	15.0	10 ~20
total	~50+15.5x3+1x10+15x20 = ~406		16 ~34

In the table, the size of replica is estimated by experience. The maximum file size per transmission is presumed to be the JPEG compressed full size picture, because text data is rather small and transmission efficiency is high (Table.2).

Still color images: (640 x 480 dots x 24bit)/
{20(JPEG compression) x 8bit} =46kB

This value is taken as 2σ and gives the average (at 50% probability) file transfer volume of 15.5kB as 0.6745σ.

We assumed here brief writing on a white board during explanation. The shared application data amount depends on control procedures rather than information quantity and we made a conservative assumption

The average volume of data per transmission is
406x8/34= 95.4 (kbit)

The average occurrence rate of burst data is estimated as $0.0188\mathrm{s}^{-1}$ which corresponds to 34 times x2(for duplex traffic) /hr, the maximum values of Table 3, in order to provide the necessary safety margins in system design. BER less than 10^{-7} will be required (probability of retransmission is 0.01 per 95.4 kbit page) and data collision rate of 10^{-4} (one retransmission per 10,000 pages) will be sufficient.

4. Capacity of Satellite Communication Channels

Here, we calculate the CDMA channel capacity of a satellite communications link under the simultaneous SCPC and CDMA transmission [1][2][3][4].

(1) Voice: SCPC channels are assigned and retained during transmission. However, there are no transmissions during silence (voice activation, or VA, is used).
(2) Video: SCPC channels are assigned and retained during transmission. VA is not used. Slow and sustained (not burst) data will be transmitted in this mode.
(3) Burst data: in the case of demand assignment, channels are assigned by access demand, and in the case of random assignment, channels are not assigned, but if the transponder load is low, CDMA transmissions are instantaneously sent, and when the load is high, they are temporarily stored in a buffer.

Next, we derive the minimum received power necessary at the earth stations to ensure the required communication quality (practically expressed in terms of BERs). Then we derive the satellite output power necessary to provide this much power at the earth stations. First, the ratio of the power received from the satellite to the thermal and interference noise must be at least as great as the down channel S/N ratio q_d required to ensure the necessary communication quality[5]. That is:

$$q_{d1} = \frac{P_{r1}/R_1}{(sP_{r3}/W + N_0)} \qquad (1)$$

$$q_{d2} = \frac{P_{r2}/R_2}{(sP_{r3}/W + N_0)} \qquad (2)$$

$$q_{d3} = \frac{P_{r3}/R_3}{\{n_1D_1P_{r1} + n_2D_2P_{r2} + (s-1)P_{r3}\}/W + N_0} \qquad (3)$$

where:

(a) q_{d1}, q_{d2} and q_{d3} are the required S/N ratios and suffixes 1, 2 and 3 are for voice, video and burst data (both here and below). However, low-speed data is assumed to be included with voice or video.
(b) P_{r1}, P_{r2} and P_{r3} are the required power per channel.
(c) R_1, R_2 and R_3 are the bit rates including error-correcting codes.
(d) D_1, D_2 and D_3 are the average duty factors.
(e) N_0 is the system noise spectrum
(f) W is the burst data spread bandwidth
(g) n_1, n_2 and n_3 are the number of (half-duplex) channels
(h) s is the number of concurrent (half) channels for burst data.

From equations (1), (2) and (3),

$$a_{11} = W/R_1, \quad a_{12} = 0, \quad a_{13} = -q_{d1}s$$
$$a_{21} = 0, \quad a_{22} = W/R_2, \quad a_{23} = -q_{d2}s$$
$$a_{31} = -q_{d3}n_1D_1, \quad a_{32} = q_{d3}n_2D_2,$$
$$a_{33} = (W/R_3) - q_{d3}(s-1) \qquad (4)$$

we have

$$\begin{bmatrix} a_{11} & a_{12} & a_{13} \\ a_{21} & a_{22} & a_{23} \\ a_{31} & a_{32} & a_{33} \end{bmatrix} \begin{bmatrix} P_{r1} \\ P_{r2} \\ P_{r3} \end{bmatrix} = N_0W \begin{bmatrix} q_{d1} \\ q_{d2} \\ q_{d3} \end{bmatrix} \qquad (5)$$

By solving the simultaneous linear equations for (4) and (5) we derive P_{r1}, P_{r2} and P_{r3}.

The equivalent isotropically radiated power (EIRP) per channel for the transponder E_{s1}, E_{s2} and E_{s3} can be derived from the equations for radio wave propagation.

$$E_{si} = L_dP_{ri}/\{\eta(\pi D/\lambda_d)^2\} \quad i = 1,2,3 \qquad (6)$$

The total average EIRP, E_{st}, is

$$E_{st} = n_1D_1E_{s1} + n_2D_2E_{s2} + sE_{s3} \qquad (7)$$

As an application example, for commercial satellites within Japan, E_s = 52.5dBW, and assuming a transponder bandwidth B_t = 36MHz and a down link frequency of 12.5GHz (wave-length λ_d =2.4cm), with forward error correction (FEC) rate 3/4, 8-valued soft decision Viterbi decoder with constraint length of 7. The communication quality (BER), bit rates, and necessary S/N ratios q_{di} are as shown in Table 4.[4]

L_d in equation (6) is put at 212.0dB including rain attenuation and 2.0dB system margin.

Table 4. Bit rates, BER and S/N Ratio

	Information source bit rate	Bit rate including FEC bits	BER	S/N (q_{di})
Voice	32kb/s	42.7kb/s	10^{-4}	2.95
Video	64kb/s	85.3kb/s	10^{-6}	3.98
Burst data	128 – 1,536kb/s	170.7 – 2,048kb/s	10^{-7}	4.57

Note: BER = Bit Error Rate

Since transponder amplification is performed within the linear operating zone, we assume a back off (headroom) condition of at least 3dB below the saturation (maximum) EIRP.

Figure 2 shows the number of concurrent channels s for burst data with a number of (half-duplex) voice channels n_1=1200 (voice bit rate of 42.7 kb/s occupies 30kHz in QPSK (Quadrature Phase-Shift Keying) modulation and all 36Mb/s transponder band width is full of voice) and video channel n_2=0 for simplicity, here.

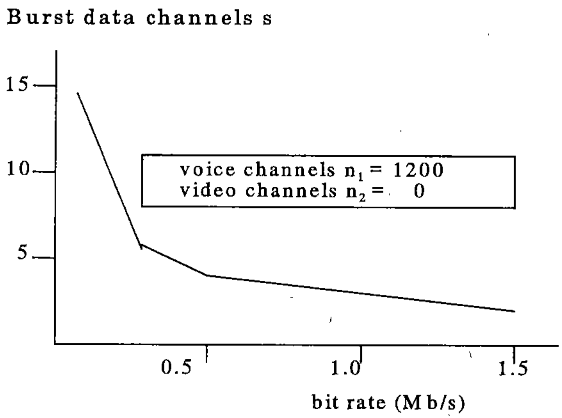


Figure 2 Number of concurrent channels of burst data

Again, burst data channels s at bit rate of 256kb/s, using the full, half and quarter transponder bandwidth are shown in Table 5, and using antenna of 1.5 and 1.2m diameter, in Table 6. The burst data capacity is roughly proportional to the transponder bandwidth and rather invariant with antenna diameter

Table5. Capacity of Burst Data Channels vs. used transponder

transponder used	1	1/2	1/4
transponder bandwidth (MHz)	36	18	9
number of voice channels	1200	600	300
burst data capacity s	7	4	2

1.8m antenna diameter

Table 6 Capacity of Burst Data Channels vs. antenna diameter

antenna diameter (m)	1.8	1.5	1.2
burst data capacity s	7	7	6

with full transponder (1200 voice channels)

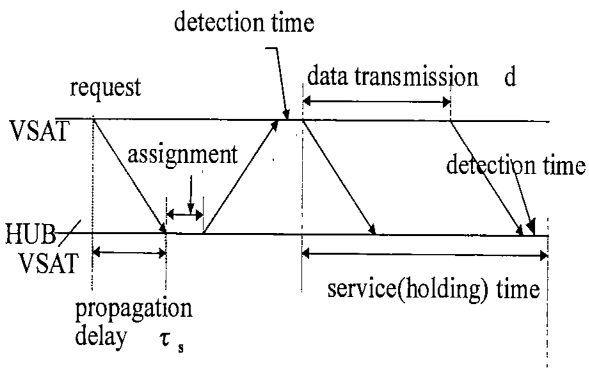


Figure 3 Time sequence of demand

5. Burst Data Transmission Capacity by Demand Assignment

A transmission request of a VSAT is sent to a hub station and then, from the hub station, the address of an open channel, if there is one, is sent to the VSAT. The station sends data on the designated channel assignment

The time sequence is shown in Figure 3. The average service time is the summation of fixed propagation delay τ_s and average data transmission time d. $h = \tau_s + d$

The time of channel allocation is estimated as

VSAT to hub(τ_s) : 0.3 (sec)
(including detection time)
Assignment processing : 0.2 (sec)
Hub to VSAT(τ_s) : 0.3 (sec)
(including detection time)

Total : 0.8 (sec)

As the probability distribution model of service time, an exponential model is selected here, and average queuing time in a VSAT buffer is stated by the following well known Erlang equations.

$$w = \frac{sB_T}{s - a(1-B_T)} \cdot \frac{h}{s-a} \tag{8}$$

and

$$B_T = \frac{a^s / s!}{\sum_{i=0}^{s} a^i / i!} \tag{9}$$

where a is the amount of traffic in Erlang, and B_T is blocking (call loss) probability without buffering of s services (circuits).

Data transmission is varied from 128kb/s to 1.5Mb/s and the throughput and total delays are shown in Figure 4. The throughput is almost flat and the delay increases in both low and high speed zones

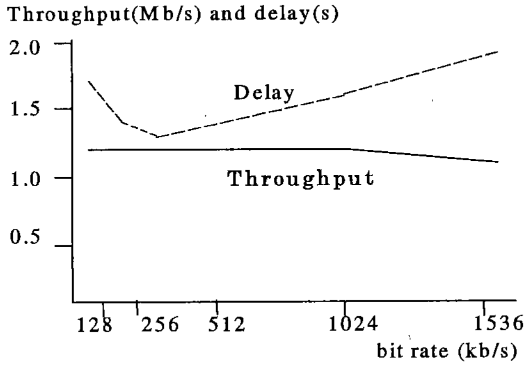


Figure 4 Throughput and delay of burst data channel (DA channels)

6. Burst Data Transmission Capacity by Random Access

In order to improve transmission characteristics, a Channel Load Sensing Protocol (CLSP) is adopted in which transmission is suppressed and the data for transmission stored in a buffer for transmission at a later time if the number of excess channels (channels not in use) is less than some specific number m. That is, when the number of data streams (number of half-circuits) exceeds the capacity r>s-m, communication quality is degraded. In order to avoid this, the earth station uses some method (discussed later) to detect r, and it suppresses transmission when r≧s-m , while accumulating the data in a buffer. Figure 5 shows the state transition diagram for the M/M/s model. In the figure, the symbols in circles represent the number of calls within the system; the symbols above and below the arrows representing state transitions are the transition probabilities.

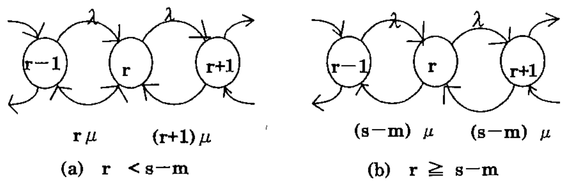


Figure 5 State transition diagram

From queuing theory[7], the probability Pr that a state is r is

$$P_r = \frac{a^r}{r!}P_0 \qquad (10)$$

when r < s-m.
And when r≧s-m,

$$P_r = \frac{a^{s-m}}{(s-m)!}\left(\frac{a}{s-m}\right)^{r-s-m}P_0 \qquad (11)$$

Here

$$P_0 = \frac{1}{\sum_{r=0}^{s-m-1}\frac{a^r}{r!}+\frac{a^{s-m}}{(s-m)!}\cdot\frac{s-m}{s-m-a}} \qquad (12)$$

The average wait time w in the buffer is

$$w = M(0)\frac{h}{s-m-a} \qquad (13)$$

Here

$$M(0) = \frac{a^{s-m}}{(s-m)!}\cdot\frac{s-m}{s-m-a}P_0 \qquad (14)$$

We calculate the probability P_c that is caused by the condition of r>s due to transmission by a number of earth stations during the delay τ_s. And τ_s is the total of the transmission delay from the satellite to the earth station and the delay to detect r. The conditions for overflow of satellite channels during the interval τ_s are that there be n +m calls, and that n -2 or fewer of the calls up till then be completed before the new call.
The probability that there exist s-m-1 calls in the satellite transponder is P_{s-m-1}, and the probability that there are n+m calls is, from the Poisson distribution,

(15)

$$P_{n+m} = \frac{(\lambda\tau_s)^{n+m}}{(n+m)!}\cdot e^{-\lambda\tau_s}$$

We calculate the probability that existing calls are completed before the new calls.
(1) Case of n=2
When zero calls are completed before the new call, satellite channel overflow occurs. The probability that, by a certain time , not even one call is completed is

$$e^{-(s-m-1)\mu\tau}$$

Here τ is random: that is, the probability is constant with respect to time, so that where the average value is R_1, then

$$R_1 \equiv \frac{1}{\tau_s}\int_0^{\tau_s} e^{-(s-m-1)\mu\tau_s} = \frac{1-e^{-(s-m-1)\mu\tau_s}}{(s-m-1)\mu\tau_s}$$

Hence the probability P_{c1} that satellite channel overflow will occur is

$$P_{c1} = P_{2+m}\cdot P_{s-m-1}\cdot R_1 \qquad (16)$$

(2) Case of n=3
If one or fewer calls is completed before the new call, satellite channel overflow occurs. The probability that the number of calls completed before the new call is zero is, as above, R_1.
The probability that one call is completed is the probability that, of s-m-1 calls, s-m-2 are not completed, and that one is completed before the new call. If the probability of the former is R_2, then this is

$$R_2 \equiv \frac{1-e^{-(s-m-2)\mu\tau_s}}{(s-m-2)\mu\tau_s}$$

whereas the latter is $\int_0^{\tau}\mu dt = \mu\tau$, and its average value is $\frac{\mu\tau_s}{2}$, so that the probability P_{c2} of satellite channel overflow is

$$P_{c2} = P_{3+m}\cdot P_{s-m-1}\{R_1 + (\frac{\mu\tau_s}{2})\cdot R_2\}$$

(17)

(3) Case of n=4

When the number of calls completed before the new call is two or fewer, satellite channel overflow occurs. The probability that the number of calls completed before the new call is zero is R_1, as in the above; and the probability that the number of completed calls is one is, again as in the preceding section,

Moreover, the probability that the number of calls completed is two is, setting

$$\frac{(\frac{\mu\tau_s}{2})\cdot R_2}{1-e^{-(s-m-2)\mu\tau_s}} \equiv R_3,$$
$$(s-m-2)\mu\tau_s$$

given by $(\frac{\mu\tau_s}{2})^2 \cdot R_3$.

Hence the probability of overflow P_{c3} of satellite channels is

$$P_{c3} = P_{4+m} \cdot P_{s-m-1}\{R_1 + (\frac{\mu\tau_s}{2})\cdot R_2 + (\frac{\mu\tau_s}{2})^2 \cdot R_3\}$$

(18)

(4) Hence, in general, we have

$$P_{c,n-1} = P_{n+m} \cdot P_{s-m-1}\{R_1 + (\frac{\mu\tau_s}{2})\cdot R_2 + \ldots + (\frac{\mu\tau_s}{2})^{n-2} \cdot R_{n-1}\} \quad (19)$$

where n≧2 and s-m≧n. That is, the probability P_c of satellite channel overflow is equal to

$$P_c = P_{c1} + P_{c2} + \ldots + P_{c,s-m-1} \quad (20)$$

Using the burst data length of 95.4 kb obtained in Section 3, and the number of simultaneous channels with burst data given the 1200 voice channels obtained in Section 4 above, we use eq. (20) to obtain the maximum transmission capacity (throughput) at which the satellite channel overflow probability is 10^{-4} or less, shown in Figure 6. Here the number of excess channels with transmission suppressed m is zero, and the bit rate is varied from 128 kb/s to 1.5 Mb/s. The average wait time w in the buffer as calculated using eq. (13) is nearly zero and the total delay time is obtained as summation of the satellite channel transmission delay and data transmission time.

Figure 7 shows the throughput when the bit rate is 256 kb/s (at which time the average service time h is 0.373 seconds for the burst data length of 95.4 kbit) and h is varied; Figure 8 shows the throughput when m is varied.

Shorter service time gives smaller throughput and m=0 is adequate where data transmission is allowed when there is at least one open channel.

7. Evaluation and Discussion

Using the equations derived in section 4, we analyze the performance of the proposed methods and their ranges of applicability. Comparing Figure 4 and Figure 6, it is clear that the Demand Assignment (DA) method offers higher throughput, but the delay is shorter for the Random Access (RA) method. Packet collisions do not occur in DA, but even in RA they can be reduced to where they do not pose a problem in actual practice.

In the DA method, the hub station must have a function for channel allocation, so that system costs are higher. Hence RA may be appropriate when burst data traffic is low, and the DA method is more suitable when burst data traffic is high.

In the RA method, burst data traffic on transponders must be detected. This may be accomplished by, for instance, the sending station notifying other stations by transmitting a tone signal corresponding to the spreading code concurrently with packet transmission. Stations subsequently wishing to transmit will then know the load on the transponder as well as the unused spreading codes.

An average page has 95.4 kbits data and the occurrence rate is $0.0188s^{-1}$ as stated in Sec.2.2, the average bit rate per terminal is 1.79 kb/s.

Therefore, the number of concurrent cooperative works supported

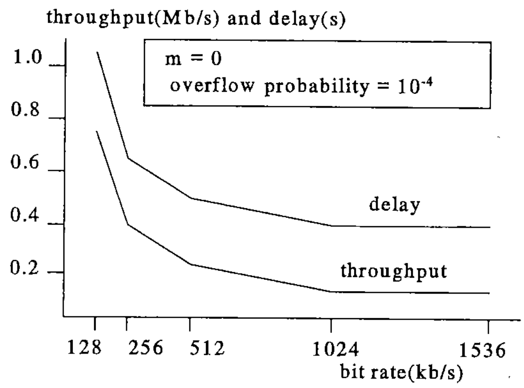


Figure6 Throughput and delay of burst data channels(RA)

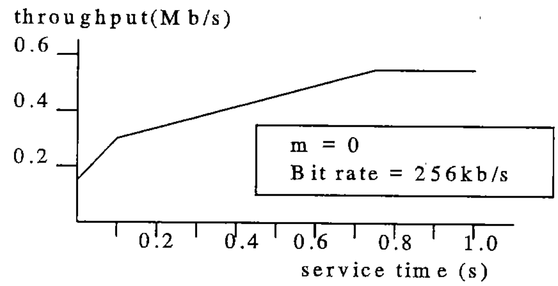


Figure 7 Burst data throughput vs. service time h

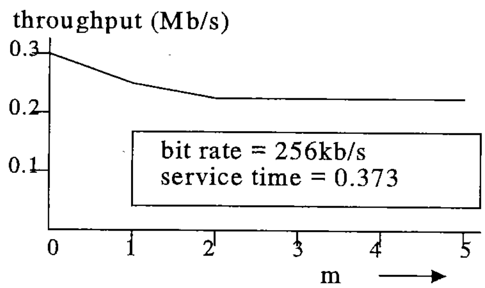


Figure 8 Burst data throughput vs. m

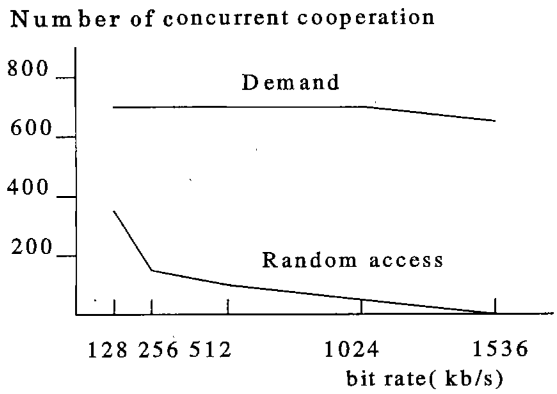


Figure 9 Supported synchronous cooperation

by a transponder in DA and RA systems are given from throughputs of Figure 4 and 6, respectively, and indicated in Figure 9.

In the case of DA, more than 600 concurrent cooperative works are supported by a transponder which is large, referring the maximum 600 duplex voice circuits (1200 half circuits).

Besides this, frequency resources of about 0.8MHz in the DA scheme and 0.1−0.5MHz in the RA scheme are saved.

8. Conclusions

We conclude that satellite communication systems with simultaneous transmission of SCPC (voice and video) and CDMA (burst data) can support many parallel distributed cooperative works without spending on extra transponder and frequency resources. This is true at least for OHP (OverHead Projector) presentation like synchronous communications where the occurrences of transmissions are infrequent and the data duration are longer.

As the access method for burst data, random access is adequate when the burst data traffic is light because of shorter delays and demand assignment for heavier traffic with slightly increased delay.

The authors gratefully acknowledge the guidance given by Professor Ryuji Kohno of Yokohama National University concerning spectral spread transmissions, and the support given by Group Manager Toshio Tachika of the Mitsubishi Electric Information Technology R&D Center.

(Manuscript received October 25,1999 , Revised manuscript received May 14, 2000)

References

[1] Miyoshi: 'A Preliminary Study on Multimedia Satellite Communication Systems', Proceedings PTC94 (1994)

[2] Miyosh: 'Optimization of Satellite Communication Systems', Ritsumeikan University (Doctor Thesis)(1994)

[3] Miyoshi, Masui: 'Spread Spectrum Multimedia Satellite Communication Systems,' Technical Report of IEICE, SAT94-60 (1994)

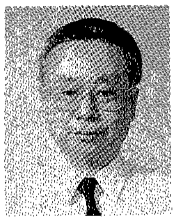
[4] Suzuki, Abe, Koizumi, Miyoshi: 'A Proposal of Satellite Communication Systems for Distributed Cooperative Work', Proceedings of 12th ICOIN (International Conference on Information Networking) (1998)

[5] Yasude, Hirata, Ogawa: 'High Rate Convolutionary Codes Suitable for Viterbi Decoding and Their Various Performances,' IECE Trans. Vol.J64-B No.7 (1991)

[6] Miyoshi, 'Performances Analysis of CDMA ALOHA Satellite Communication Systems', Research Report of Hiroshima-denki Institute of Technology Vol.30 (1997)

[7] Akizuki, Kawashima: 'Information and Communication Traffic--Fundamentals and Applications,' Ohm Publishing Co. (1990)

Masanobu Suzuki (Non-member) received the B.E. degree in mathematics in 1969 from Tohoku University. He jointed Mitsubishi Electric Corporation in 1969. He has been engaged in research and development of operating systems, software production engineering, information technology and multimedia communication technology. He is a member of the IPSJ.



Nobuhiro Kataoka (Non-member) received the B.E. and the M.E. degree in electric engineering from Osaka University in 1966 and 1968 respectively, and Ph.D. degree in computer science in 2000 from Tohoku University. He jointed Mitsubishi Electric Corporation in 1968. He has been engaged in research and development of operating systems, software engineering, and computer communication system design. His research interests include ERP packages and Intranet as the productivity tools for office workers. Since April 2000, he is a Professor at the Department of Engineering of Tokai University. He is a member of the IEEE, IPSJ, IEICE, and the Japan Society of Management Information.

Kazuyoshi Miyoshi (Non-member) received the B.E. degree in electrical engineering from Kyoto University in 1957 and the Ph.D. in communication engineering from Ritsumeikan University in 1994. He joined Mitsubishi Electric Corporation in 1957. He has been engaged in research and development of communication satellites and in systems engineering of satellite communication systems. Since April 1993, he is a Professor at the Department of Computer Science, Hiroshima Kokusai Gakuin University. He is a member of the IEEE, IEICE, Japan Society for Aeronautical and Space Sciences and the Operations Research Society of Japan.

Hisao Koizumi (Mmember) received the B.E. degree in communication engineering in 1961 and the Ph.D. degree in computer science in 1996, both from Tohoku University. He jointed Mitsubishi Electric Corporation in 1961. He has been engaged in research and development of operating systems, software engineering, and computer communication system design. Since April 1998, he is a Professor at the Department of Computer and System Engineering of Tokyo Denki University. He is a member of the IEEE, IPSJ, IEICE, and the Japan Society of Mechanical Engineers.