VOICE TRANSMISSION THROUGH THE CODE DIVISION MULTIPLE ACCESS LOCAL AREA NETWORK

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ABSTRACT

In this paper the performance of the code division multiple access (CDMA) local area network (LAN) has been investigated by computer simulation when the network is used for transmission of voice as well as data. For this purpose, several voice coding methods have been tested using real speech.

Two cases have been simulated. In one case, voice is transmitted through a constant-rate link. The pulse code modulation (PCM), pulse amplitude modulation (PAM) and hybrid companding delta modulation (HCDM) coding methods were tested. The results show that PAM gives better quality than PCM when multi-level (i.e., m-ary) data symbols are used, and that the adaptive quantization scheme greatly improves the quality of the received speech.

In the other case of voice transmission, voice is transmitted through a variable-rate link. We used embedded (or variable-rate) coding methods such as PCM with the bit-stripping scheme and the residual-excited linear prediction (RELP) vocoder with an embedded structure. These coding schemes yield ideal performance in the sense that the voice quality is degraded slowly as the network becomes loaded.

I. INTRODUCTION

So far, voice signal has normally been transmitted through a circuit-switched network, whereas the transmission of data has been made through a packet-switched network. Recently, however, the idea of integrating voice and data has emerged, and has been a topic of increasing interest. The major reason for integrated voice/data transmission is the expected cost savings obtained by sharing transmission and switching facilities. Also, integration of voice and data facilitates the design and implementation of services pertaining to both voice and data communication, such as the electronic voice mail.

It is well known that voice and data have different characteristics and different transmission requirements. For example, voice packets are generated at a constant rate while a person is talking. Delivery of these packets must be made within a maximum tolerable delay. The loss of 1-2 percent of all packets does not seriously degrade the voice quality [1]. A voice call may occasionally be blocked. However, once it is established, it must be kept uninterrupted. On the other hand, data must be transmitted with a very low error rate. Every bit of the data is considered to be significant. However, delay is somewhat tolerable for data. A delay of as much as 100 or 200 ms is quite common in many applications.

The approach to intrgrating voice and data traffics frequently used in packet-switched networks or in synchronous time division multiplexing (STDM) systems is to divide a frame (or slot) into two subframes and allow the two traffics to be serviced in a different manner [2]. For example, in a STDM system, voice and data occupy a circuit-switched area and a packet-switched area, respectively. The boundary of the two subframes may be fixed or movable depending on the protocol used. Also, the voice subframe is normally given priority to reduce the transmission delay. A similar approach can also be found in some local area networks [3].

Several researchers recently investigated the performance of the carrier sense multiple access with collision detection (CSMA/CD) local area network (LAN) [4] or a unidirectional-bus LAN [5] in voice/data integrated environments by computer simulation. According to their results, combining voice and data traffics on a LAN renders the system economical and flexible. However, integrating voice and data on a conventional network requires an elaborate approach and schemes, such as packet voice formatting, synchronization, time stamping, silence detection and regeneration, and buffering and reassembling of voice packets [6]. Such problems can be eliminated by integrating voice and data on the code division multiple access LAN [7],[8].

In this paper we investigate the performance of the CDMA LAN in a voice/data integrated environment by computer simulation. Following this introduction, in Section II a review of the CDMA LAN proposed in [7],[8] is given. In Section III computer simulation results are presented and discussed. Finally, conclusions are made in Section IV.

II. CODE DIVISION MULTIPLE ACCESS LOCAL AREA NETWORK (CDMA LAN)

In this section we briefly review the code division multiple access (CDMA) local area network (LAN) proposed in our earlier papers [7],[8]. The basic configuration of a CDMA LAN is shown in Fig. 1. It is basically a broadcasting network with a dual-cable

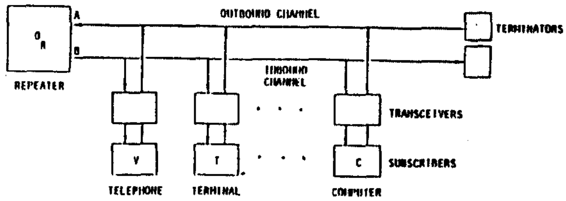

Fig. 1 Configuration of a basic CDMA LAN.

configuration. Each transceiver is connected to two coaxial cables, one for inbound and the other for outbound data, by using taps. The repeater in Fig. 1 is simply an analog amplifier. The signal at the point A is amplified by the repeater and then broadcast at the point B on the inbound channel (cable) together with a reference sequence. The reference sequence is used to synchronize all the transceivers in the network.

The principle of signal separation by the CDMA method is based on the orthogonal properties that may be found in some orthogonal sequences [9]. In the CDMA LAN we use the bit-shifted maximal-length sequences as orthogonal codes. N different data can be recovered from a symbol by using the method of correlation detection, N being the period of the maximal-length sequence. The symbol is formed by summing all individual symbols which are the bit-shifted maximal-length sequences multiplied by the user data. Thus, a total of N independent channels are available in the CDMA LAN. They are referred to as the logical channels of the CDMA LAN. Because each logical channel uses a bit-shifted maximal-length sequence, each logical channel is distinguished by the amount of bit shifting.

One aspect that is noteworthy is that we use multi-level symbols representing m-ary data in order to handle traffics of different data rates and error rates. For example, a user who transmits interactive data may require a low error rate, while the transmission rate is relatively low. On the other hand, a user who transmits voice can have a higher transmission rate than that of data.

Multiple access in the CDMA LAN is done according to a distributed reservation protocol which is based on the dynamic allocation of logical channels to the users. The gist is basically the following. In the CDMA LAN a total of N channels are provided by the system. One of them is used as the reference channel for broadcasting the reference data pattern on the reference sequence (i.e., the original maximal-length sequence). Several other channels are used as common control channels referred to as the reservation channel. The rest are used as user channels. We slice time into slots, each of which consists of a fixed number of consecutive data bits. The starting bit of the slot is identified by using the reference data pattern serviced by the repeater. All the users who would use the medium (i.e., user channels) at the next slot must reserve it by using a bit map method. That is, if a station (user) has any message to transmit, it writes a 1 bit into its bit location on the reservation channel. After all the addresses of the applicants have been known at the end of the current slot, the user channels are equally divided in number among the stations which have made the reservations. If a station has no message to transmit, it makes no reservation, thus becoming disconnected from service. As the load (i.e., the number of active stations) grows, the average number of channels occupied by a station will decrease until each station utilizes only one channel at full load.

We now briefly describe three reservation methods used in the CDMA LAN [8]. We assume that the propagation delay is a_c times the duration of a slot. Initially, the station is in the idle state. In that state, it can receive a message generated by the user. On receiving a message, it reserves the medium on the reservation channel. The station which has applied for the medium access on the reservation channel receives permission to access the medium after a_c slots from its reservation. Thus, the user channels can be allocated and used after (a_c+1) slots from the first reservation slot. Then, the station transmits the message on the allocated user channels over a number of slots. While transmitting at each slot, the station determines whether it will keep on reserving the medium by checking the length of the message that will be left after the current transmission. In this way, the station reserves the medium continuously until it finds that no message is left. This reservation method (called the method A in this work) is simple and straightforward. However, since a_c slots have already been reserved, the user channels allocated at those a_c slots are discarded.

To reduce the number of discarded slots, we consider a different reservation method (called the method B in this work). In this method, the station reserves the medium only for ℓ_0 consecutive slots when the length of the generated message ℓ_0 is shorter than or equal to (a_c+1) slots. Also, the station always counts the number of reserved slots NRSRVD, which is incremented and decremented with reservation and transmission at each slot, respectively. If the length of the remaining message is shorter than or equal to NRSRVD, it ceases reservation. Thus, this method reduces the number of discarded slots as compared with the method A.

The method B can also be modified to give improved utilization. In this method (called the method C in this work) the station reserves only one channel at each slot for the case that $\ell_0 \leq a_c + 1$.† Such single-channel reservation requires another reservation channel. The station which has reserved the medium on this channel can occupy only one channel at the next slot. Such a single-channel reservation scheme enables the user to have a circuit on which data can be transmitted at a constant rate without being disturbed by other users. Thus, transmission of voice can be facilitated by this method.

In the CDMA LAN the transmission rate of a data link is changed slot by slot, when the normal reservation method (i.e., the method A or B) of the CDMA LAN is used. Although the data rate is not constant, it is always a multiple of the rate of a logical channel. Therefore, in this case a voice coder with an embedded structure such as pulse code modulation (PCM) and residual-excited linear prediction (RELP) coders would be appropriate. This enables the transmitting station in the CDMA LAN to adjust the trans-

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† This method is the same as the method B for $\ell_0 > a_c+1$.

mission rate of a voice coder according to the number of allocated logical channels at each slot.

III. COMPUTER SIMULATION RESULTS AND DISCUSSION

A block diagram for the simulation of voice transmission over the CDMA LAN is shown in Fig. 2. As seen in the figure, it has been assumed in our simulation that only one voice user is transmitting a long voice message to another voice user through the CDMA LAN. Consequently, only one data link is used for voice transmission. All other channels (or links) are occupied by data users who exchange interactive data. The interactive data messages are generated by using a random number generator. The length of the generated message is geometrically distributed. On the other hand, the voice message is read from a speech file. It is processed by a voice coder, and forwarded to the CDMA LAN simulation program which performs the multiaccess operation discussed in Section II. The reservation methods B and C discussed in Section II were used for transmitting the encoded speech through the normal and single-channel links, respectively. Unlike other LAN's based on time division multiple access (TDMA), the encoded speech is directly transmitted through the established voice link without the need of packetization or buffering. When the transmitted voice is received by the destination station, it is decoded and written on an output file.

For simulation of voice transmission over the single-channel data link (i.e., the reservation method C discussed in Section II), we used pulse code modulation (PCM), pulse amplitude modulation (PAM) and hybrid companding delta modulation [10] coding methods. In PCM or HCDM coding, a number of consecutive output bits of the voice coder are grouped and treated as a binary number, and then transmitted on a multi-level symbol. On the other hand, in PAM coding the coder output bits are not significant as a binary number but changed directly to a corresponding symbol. In this case, the interference and channel noise in the CDMA LAN can be regarded as input noise to the voice coder if the voice input is quantized uniformly. For this reason, we tested a uniform quantizer for PAM and PCM. Also, we tested PAM with μ-law companding and adaptive quantization schemes [11].

On the other hand, for simulation of voice transmission over the normal CDMA data link (i.e., the reservation method B), we used the PCM coder with a bit-stripping scheme and the residual-excited linear prediction (RELP) [12] vocoder with an embedded structure. In this case, only one bit is always transmitted per symbol, that is, $q=1$. In our simulation, the residual signal of the RELP vocoder was divided in the frequency domain into 7 "baseband" signals. Consequently, the bandwidth of each baseband is about 571 Hz. The baseband signals are separately coded by using the adaptive PCM (APCM) scheme discussed before. At the receiver part, they are decoded and added to yield the excitation signal. When the network is lightly loaded, all the baseband signals are transmitted to the receiver. However, when the traffic in the network is heavy, a number of uppermost basebands cannot be transmitted. In this case, the receiver fills the empty basebands with the conventional linear predictive coding (LPC) excitation signal with the same band frequencies. Thus, the

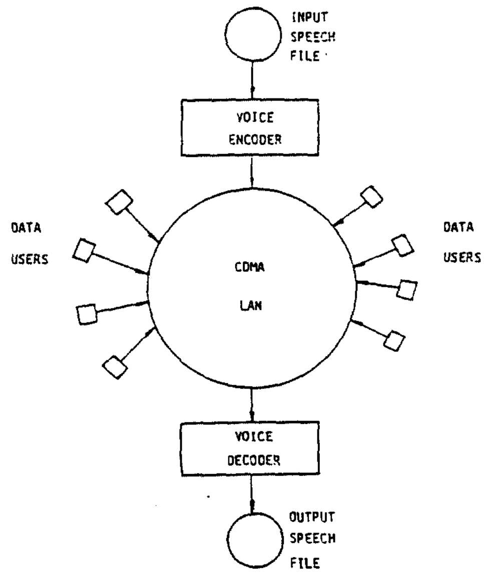

Fig. 2 Block diagram for the simulation of voice transmission over the CDMA LAN.

embedded RELP vocoder is reduced to the conventional LPC vocoder when the network is fully loaded, that is, when only one channel is available.

It was assumed in our simulation that 1020 user channels were available out of 1023 channels. The number of subscribers was one half of that of user channels, that is, $Z=510$. The data generation rate in the idle state and the mean length of data messages were 0.9 and 4 slots, respectively. The effect of propagation delay was assumed to be negligible.

First, voice was transmitted through a single logical channel on multi-level symbols. The medium data rate was assumed to be 10 Mbits/s. In this case, the data rate of a single-channel link is about 9.8 kbits/s (for binary data). Fig. 3 shows the

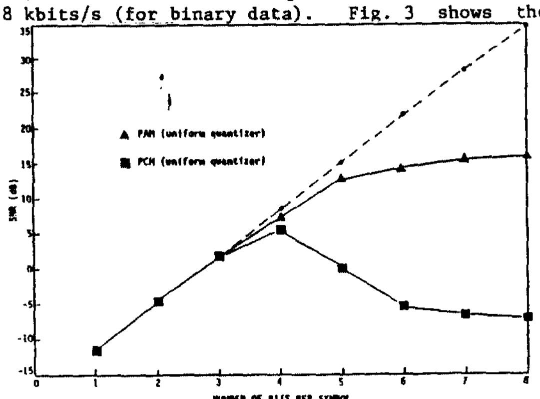

Fig. 3 SNR versus the number of bits sent per symbol when voice is transmitted by PAM and PCM over the CDMA LAN. [The dotted line gives performance of PCM obtained when coded voice is directly decoded without being transmitted through the CDMA LAN]

simulation results when the PCM and PAM coding methods were used. According to the simulation result, the PCM coding shows rather poor performance as compared with the PAM coding scheme. When the number of bits per symbol, q, is small (i.e., q=1~4), the SNR obtained for PCM coding is low because of quantization noise. Since the error rate of the CDMA LAN is low for a small value of q, the performance follows the one obtained without transmitting the voice through the LAN. For a large value of q, however, the SNR continues to decrease, and approaches to a low value of SNR. The PAM coding scheme shows performance which is somewhat similar to that of PCM coding for a small value of q. As q becomes large, the PAM coding simply approaches analog (although discrete in time) transmission. Therefore, the SNR obtained for PAM coding is bounded by a value determined by the interference and Gaussian noise of the CDMA LAN. Thus, a large gain in performance can be obtained by using the PAM scheme over the normal PCM coding when voice is transmitted on multi-level symbols in the CDMA LAN. This can be achieved simply by changing the PCM codeword into the PAM codeword.

Figs. 4 and 5 show the results obtained for PAM coding with μ-law companding (μ=255) and adaptive schemes, respectively. The μ-law companding does not improve SNR when the number of bits per symbol is large. For a small value of q, however, the μ-law companding is advantageous compared with the result for the uniform quantizer. Thus, the μ-law coding scheme is shown to widen the dynamic range of the coder. Fig. 5 shows that the adaptive quantizer greatly improves the coder performance. For a large value of q, however, the performance is shown to be degraded. The reason for this is due to the fact that its performance is rather sensitive to channel errors because the adaptive quantizer has a 1-word memory. This effect can also be seen in Fig. 6.

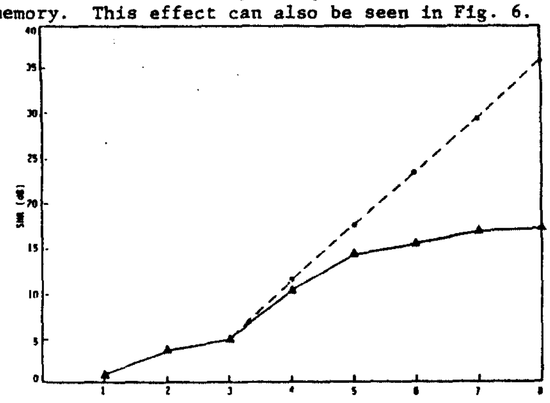

Fig. 4 SNR versus the number of bits sent per symbol when voice is transmitted by μ-law PAM over the CDMA LAN. [The dotted line gives performance of μ-law PAM obtained when coded voice is directly decoded without being transmitted through the CDMA LAN] (μ=255)

The results of HCDM coding for q=2,3 and 4 are listed in Table I. They correspond to transmission rates of 19.5, 29.3 and 39.1 Kbits/s, respectively. For these values of q, the channel error rate is very low, as can be seen in Fig. 3. Therefore, the SNR's obtained for q=2, 3 and 4 are very close to those obtained for the case that voice is decoded directly

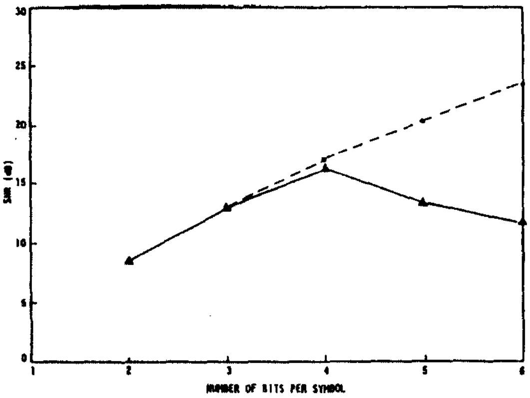

Fig. 5 SNR versus the number of bits sent per symbol when voice is transmitted by adaptive PAM over the CDMA LAN. [The dotted line gives performance of adaptive PAM obtained when coded voice is directly decoded without being transmitted through the CDMA LAN]

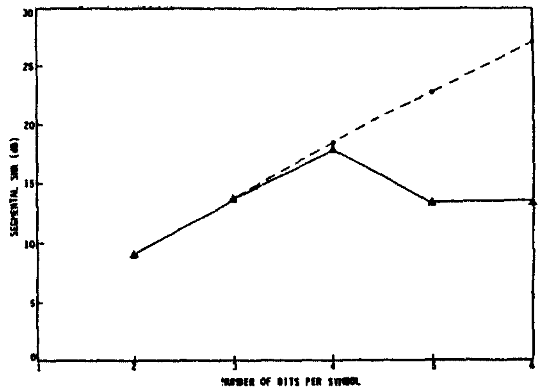

Fig. 6 Segmental SNR versus the number of bits sent per symbol when voice is transmitted by adaptive PAM over the CDMA LAN. [The dotted line gives performance of adaptive PAM obtained when coded voice is directly decoded without being transmitted through the CDMA LAN]

Table I. Simulation results obtained for the HCDM coding.

Number of bits per symbol (q)	Without channel errors†		With channel errors††	
	SNR	SNR_{SEG}	SNR	SNR_{SEG}
2	16.1	15.1	16.1	15.1
3	21.4	20.7	21.1	20.5
4	25.9	24.2	25.2	24.0

† Simulation results obtained when HCDM-coded voice was directly decoded without being transmitted through the CDMA LAN.

†† Simulation results obtained by transmitting the coded voice through the CDMA LAN.

without being transmitted through the CDMA LAN.

We next discuss the performance of the PCM and RELP voice coding methods for the case that voice is transmitted on a normal CDMA LAN data link. In this case, the coders use multiple channels on which binary data (i.e., q=1) are transmitted. It was assumed that the maximum number of channels that a user can use was 8. Fig. 7 shows the performance of a PCM coder with the bit stripping scheme as a function of the number of subscribers. The medium data rate was assumed to be 10 Mbits/s. As seen in the figure, the performance is satisfactory in that it is degraded almost linearly with the increase of the number of subscribers. When the number of subscribers is very small (say, less than 300), each user can have 8 channels which is the maximum number allowed. Therefore, the SNR is not increased over some value for small values of q.

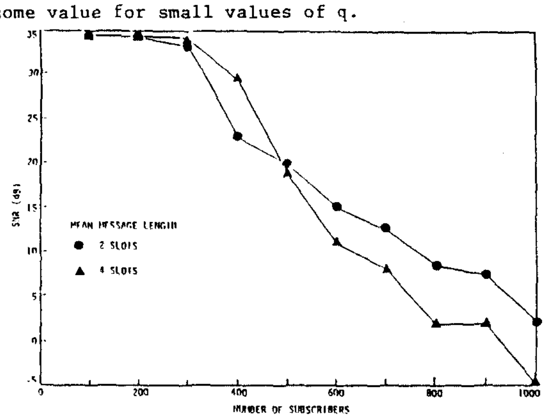

Fig. 7 SNR versus the number of subscribers for mean message lengths of 2 and 4 slots when voice is transmitted by embedded PCM coding over the CDMA LAN.

Lastly, let us discuss the results obtained for the RELP vocoding method. The medium data rate was assumed to be 2.35 Mbits/s. Hence, the data rate of a channel is about 2.3 Kbits/s. When only one channel is allocated, the system operates as a standard LPC vocoder. When more than one channel is allocated, each of the additional channels is used for transmitting the baseband residual signal. The number of samples in an analysis frame was assumed to be 175. Each sample is quantized by a 2-bit APCM coder. Consequently, two frames of baseband residual samples are included in a slot, which is assumed to consist of 100 symbols (or bits). According to our informal listening test, the quality of the output speech was judged to be quite clean when the number of subscribers was relatively small. However, the buzzy effect, which is characteristic of the LPC vocoder, began to be heard as Z became large. At full load (i.e., Z=1020), the output speech quality was found to be the same as that of the LPC vocoder.

IV. CONCLUSION

In this paper, voice transmission over the CDMA LAN was investigated by computer simulation. The CDMA LAN is very suitable for integration of voice and data traffics. In the simulation, real speech was transmitted through one voice link. All other links were assumed to be occupied by data users.

We simulated two cases. In one case, voice was transmitted through a constant-rate link. Several voice coding methods, such as PCM, PAM and HCDM, were tested. We used PCM with a uniform quantizer. Also, we used three schemes for PAM coding; uniform quantization, μ-law companding and adaptive quantization schemes. The results show that PCM gives better quality than PCM when multi-level data symbols are used, and that the adaptive quantization scheme greatly improves the quality of the received speech. The PCM-to-PAM conversion can be done simply by changning the codeword received from the user.

In the other case of voice transmission, voice was transmitted through a variable-rate link, rather than a constant-rate link on which multi-level (m-ary) symbols were used. In this case, the number of logical channels was varied slot by slot. We used embedded (or variable-rate) voice coding methods such as the PCM with the bit-stripping scheme and the residual-excited linear prediction (RELP) vocoder with an embedded structure. These coding schemes show ideal performance in the sense that the voice quality is degraded slowly as the network becomes loaded.

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