RC 20838 (92275) 7MAY97
Engineering and Technology 14 pages

97A000412

Research Report

Voice/Data Traffic Management in the Return Path of Two-Way Hybrid-Optic Coaxial Networks

by: Te-Kai Liu
IBM Research Division
T.J. Watson Research Center
P.O. Box 218
Yorktown Heights, NY 10598

IBM **Research Division**
Almaden · T.J. Watson · Tokyo · Zurich · Austin

Voice/Data Traffic Management in the Return Path of Two-way Hybrid Fiber-optic Coaxial Networks

Te-Kai Liu
IBM Thomas J. Watson Research Center
30 Saw Mill River Road
Hawthorne, NY 10532-1507
(914) 784-7361
tekailiu@watson.ibm.com

Abstract – Two-way hybrid fiber-optic coaxial (HFC) networks have enabled the integrated services of high-speed Web surfing, additional phone lines and video on demand etc. on top of the existing CATV broadcast service. However, the key to the success of this technology relies on the quality of service (QoS) provided to the subscribers. In terms of the available bandwidth and channel quality of the two-way transmission system, the reverse path (from subscribers to the headend) is most likely the problematic factor toward subscriber satisfaction due to its limited bandwidth (5-42 MHz) and inherent noise characteristics in the tree-and-branch network topology. This paper investigates some traffic management issues in order to guarantee the diverse QoS of different traffic types in the return path of the HFC network where code division multiple access (CDMA) technique is employed as the medium access protocol. In particular, the tradeoff between the QoS of voice and data services, and strategies for the headend to improve the QoS of the integrated traffic are studied and quantified by numerical examples.

1. Introduction

The capability of transmission on the return path of a two-way HFC network has enabled many interactive services, such as video on demand, extra phone lines, telecomuting, Web surfing, and so on. The tremendous bandwidth on the forward (broadcast) path of HFC networks dramatically reduces the time for downloading multimedia information from the network and makes Web surfing over HFC networks more enjoyable than over traditional telephone lines. Although the benefits are great, this can only be achieved if the return path also provides high quality service, which is not so straightforward. In a two-way CATV system, the return path operates in the 5-42 MHz band which is subject to various ingress noise coming from shortwave broadcast, amateur radio, citizen band radio, and impulse noise due to motor ignition and electric appliances etc. [1], [2]. The diverse noise sources are further funneled to the headend because of the tree-and-branch topology of the network.

Various approaches have been proposed to reduce the aggregate noise at the headend [3], [4]. For examples, reducing the number of subscribers per fiber node (the interface of coaxial and fiber-optic trunks) can reduce noise funneling effect. Also, improving the quality of installation at subscriber sites and periodic diagnosis of cable plants can cut down the possibility of ingress noise entering the return path. Moreover, the frequency agility ability of cable modems can be used to get around the noisy channels when narrowband ingress noise occurs. The above approaches aim to cope with noise in the cable plant at the price of more equipments (e.g., fiber nodes) or higher maintenance labor cost.

Spread spectrum technique also known as CDMA (code division multiple access) [5] offers another way to handle the noise problem in HFC networks [6]. With the processing gain coming from bandwidth expansion of the transmission signals, the headend is able to detect the transmission from subscribers in a much more harsh environment. but the bandwidth efficiency is not as good as other multiple access techniques such as FDMA or TDMA in a friendly environment, i.e., the return path is not very noisy. The lower bandwidth efficiency of CDMA motivates our study of traffic management strategies which are aimed to make the most use of the available spectrum on the return path so as to deliver high quality service (low packet error rate and short access delay etc.) over a possibly noisy return path.

In this paper we develop an analytical model to study the QoS of voice and data traffic in a shared CDMA return channel. The results for single channel are further extended to a multi-channel scenario where traffic management strategies are evaluated. The rest of the paper is organized as follows. Section 2 introduce the system model and the QoS measures for voice and data services. Section 3 present numerical results and show the tradeoff between the QoS of voice and data traffic for a single channel. Section 4 considers traffic management issues in the setting of multi-channel environment. Some concluding remarks are given in Section 5.

2. System Model

The considered CATV network consists of a headend at the root of a tree-and-branch distribution network which provides both voice and data services for a large number of subscribers in the service area. Among the various types of traffic on the return path of the network, 2 distinct types are identified, namely the constant bit rate (CBR) traffic and random access traffic. The former comprises of voice conversation and file transfer such as FAX transmission. Typically, CBR traffic is serviced by first establishing a connection and get the required bandwidth reservation from the headend. The signalling messages are usually short in length and arrive at the headend in a random fashion. Moreover, one of the killer applications in a two-way CATV system is Web browsing where users generate requests for multimedia information by mouse clicking. The request messages and other handshaking messages generated at the client (subscriber) sides are typically short and random. Therefore, we model the traffic on the return path as a combination of voice-like CBR connection and Poisson type of data traffic.

We assume the headend provides the required synchronization signals so that all subscriber cable modems can synchronize their packet transmission at the slot boundary where a slot duration is equal to the transmission time of a fixed-size packet. The idea of fixed-size packet eases the hardware implementation and can efficiently meets the requirement of ATM standard. In case the message to be sent out is longer than can be accommodated in a single packet, message fragmentation and reassembly are done by a higher layer protocol, e.g., TCP/IP.

Packet transmission using direct sequence CDMA is considered in this study. Due to the static topology of a cable network, power control is easier to do than in the wireless cellular world. With the standard Gaussian assumption for other-user interference and background noise, the effective signal to noise ratio E_b/N_{eff} is given by $\frac{E_b}{N_{eff}} = \left[\left(\frac{E_b}{N_0}\right)^{-1} + \left(\frac{G}{k}\right)^{-1}\right]^{-1}$, where E_b/N_0 is the signal to noise ratio due to thermal and ingress noise[1], G is the processing gain, and k is the number of simultaneous interfering transmission. Depending on the modulation scheme, the instantaneous bit error rate can be computed accordingly. For example, $BER(k) = \frac{1}{2}\exp\left(-E_b/N_{eff}\right)$ for DPSK modulation with hard decision decoding.

Since the connection time of voice service is much longer than a packet transmission time, we consider a system that M_v voice connections are being made and the aggregate data traffic

---

[1] Although ingress noise usually comes in the form of narrowband interference, it will look like a broadband noise after being despread at the headend receiver.

constitute an average arrival rate of λ_d packets per slot with the Poisson distribution.

Define BER(m, λ_d) to be the average bit error rate of a test packet when there are m interfering voice users and Poisson data traffic with an average load of λ_d packets/slot. BER(m, λ_d) can be computed by

$$BER(m, \lambda_d) = \sum_{j=0}^{\infty} BER(m+j) \frac{e^{-\lambda_d} \lambda_d^j}{j!}.$$

Suppose the test packet is encoded by BCH(n, k, t) forward error correction codes [7] (n is the packet size in bits, k is the number payload bits, and t is the number of correctable bit errors), the average packet error rate (PER) given m interfering voice users and mean data traffic of λ_d packets/slot is given by

$$PER(t, m, \lambda_d) = 1 - \sum_{i=0}^{t} \binom{n}{i} BER^i (1 - BER)^{n-i}.$$

The PER experienced by a voice user (the QoS measure for voice traffic) depends on the voice code rate r_v ($= k/n$), the interfering voice traffic load m ($= M_v - 1$), and the data traffic load λ_d. As for the data service, we define the QoS measure as the net data throughput which is given by

$$S_d = \lambda_d \cdot r_d \cdot [1 - PER_d(r_d, m, \lambda_d)] \quad \text{(packets/slot)},$$

where the code rate of data packets r_d excludes the overhead due to the FEC codes from the calculation of the net data throughput, which has a unit of packets/slot. In the numerical results presented in the next section, we show how the choices of r_v, r_d and λ_d influence the QoS of voice and data services.

3. Analytical Results for Single Channel

The parameters used in this study is given in Table 1. Depending on the bit rate of a vocoder R_b bps, the processing gain $G = 128$ makes the spread signals occupy a bandwidth of R_b*G Hz. For example, $R_b = 32$ Kbps corresponds to a spread bandwidth of 4.096 MHz. The choice of packet size = 511 bits is to accommodate a ATM cell plus the overhead due to BCH(511, 430, 9) FEC code.

Table 1: System Parameters

Item	Symbol	Value
Processing gain	G	128
Packet size (bits)	n	511
Number of correctable bit errors for voice packets	t	9
Number of correctable bit errors for data packets	t	3, 6, 9
Bit energy to noise spectral density (dB)	E_b/N_0	5 ~ 30
Number of voice connections	M_v	0 ~ 31
Data traffic load (packets/slot)	λ_d	0 ~ 31

Fig. 1 plots voice PER versus E_b/N_0 with 11 voice connections and data traffic load ranging from 5 to 15 packets/slot. Suppose the voice QoS requires that PER is at most 10^{-3}, the intersection of the horizontal line with the PER curves show the minimum E_b/N_0 (or the strongest funneled noise) at the headend receiver. As expected, higher noise power (lower E_b/N_0) results in lower supportable data traffic. However an excessive data traffic load ($\lambda_d = 15$ in this case) will make the voice QoS unacceptable, which means that data traffic load needs to be controlled.

Fig. 2 shows the voice PER versus the controlled data traffic load for various numbers of existing voice connections when the noise is negligible[1]. The corresponding data throughput is given in Fig. 3, which says that there is an optimal data load for a given M_v even without the constraint of voice PER. We will see later that the system cannot operate at the point of peak data throughput due to the constraint of voice PER. The tradeoff between voice PER and data throughput is shown in Fig. 4 where the difference between the supportable data throughputs with or without voice PER constraint can be seen clearly.

The effect of different data coding rate is plotted in Fig. 5. For voice PER between 10^{-3} and 10^{-2} and $M_v = 11$, data code rate = 0.89 (the number of correctable bit errors $t = 6$) gives the highest throughput.

Fig. 6 shows the maximum supportable data throughput versus the number of voice connections under 2 different voice PER requirements. As expected, the supportable data throughput decreases as M_v increases. When there is no voice connection, however, the data traffic load can be driven higher to get the data throughput equal to 23.18 ($\lambda_d = 31$ in this case.) This suggests that it is better off to separate data traffic from voice traffic as far as the data capacity is concerned. Nonetheless, we will see later that there exist reasons that it is advantageous to mix voice and data traffic in a single channel.

Fig. 7 plots system efficiency versus M_v for different data code rates, where the efficiency is defined to be the ratio of the sum of data throughput and M_v to the maximum number of supportable voice connections[2] under the constraint of voice PER = 10^{-2}. The results indicates that the bursty nature of data traffic has a negative effect to system efficiency.

---

1. The noise effect will be neglected in the rest of the numerical results.
2. The maximum number of supportable voice connections is 31 in this case.

4. Traffic Management in a Multi-channel Environment

Having demonstrated the QoS tradeoff of voice and data traffic in a single channel, we would like to discuss the strategies for traffic management when there are multiple CDMA channels, i.e., in different frequency bands.

From the results in section 3, one knows that in the heavy traffic scenario where system is fully loaded with voice connections and data traffic, it is more efficient to allocate voice connections in channels without random data traffic. Similarly it is also better off to segregate voice service from channels that are supporting data traffic, since data channel without voice PER constraint can achieve higher throughput. But the question is "how to allocate voice and data traffic when the system (with multiple channels) is not fully loaded? "

When the system is not fully loaded, the requirement for voice service can be kept the same but data traffic would probably care more about the retransmission delay when data packets are in error rather than data throughput alone[1]. To show how the data service can be improved by properly off loading some traffic to the voice channel, we consider the following example.

Suppose the system has 2 independent CDMA channels, each can support up to 31 voice connections (the same as the system given in Table 1.) At a particular point of time, there are M_v (< 31) voice connections being made in one of the 2 independent CDMA channel and a total data traffic load of $\lambda_d = 30$ packets/slot. How should the headend properly split the data load into the 2 channels so that data PER is minimized while the voice PER is still maintained above 10^{-3}?

Fig. 8 shows the average data PER of the 2 channels as a function of data load allocated to the voice channel which has M_v ongoing voice connections. It can be seen that there exists an optimal

---

[1] The data throughput will be very close to the offered load when the traffic load is light.

split of the total data traffic which will minimize the data PRE for a given M_v. The choice of the amount of data traffic allocated to the voice channel needs to be constrained by the voice PER requirement. Fig. 9 plots the voice/data PER as a function of the data load in the voice channel. We see that when $M_v = 11$, the optimal shifted data load is not affected by the voice PER constraint, whereas when $M_v = 26$, the voice PER constraint does not allow data traffic to operate at the point of the lowest PER. But suffice this, average data PER is still reduced from 0.08 to 0.05, an improvement of 35%. The improvement will be experienced by subscribers in the form of shorter access delay in the return path.

5. Conclusion

The QoS tradeoff and some traffic management issues in providing integrated voice/data services on the return path of two-way HFC networks employing CDMA technique are investigated. The numerical results show that when the system is fully loaded with voice/data traffic, it is more efficient to segregate CBR voice service from random data packets in the return path. Otherwise, the data PER (and thus the total access delay) can be improved by off loading some amount of data traffic to the voice channel which is currently underutilized.

References

[1] C. A. Eldering, N. Himayat, and F. M. Gardner, "CATV Return Path Characterization for Reliable Communications," *IEEE Communication Magazine*, Aug. 1995, pp. 62-69.

[2] R. P. C. Wolters, "Characteristics of Upstream Channel Noise in CATV-networks," *IEEE Trans. on Broadcasting*, vol. 42, no. 4, Dec. 1996, pp. 328-332.

[3] K. J. Oliver, "Preventing ingress in the return path (CATV)," *CED* (USA) vol. 22, no. 11, Oct. 1996, pp. 36-8, 40.

[4] A. Mends-Cole, "Operational Issues in a Two-way HFC Environment," *Proceedings of SPIE*, vol. 2609, 1995, pp. 60-66.

[5] M. K. Simon et al., *Spread Spectrum Communications*, Computer Science Press, 1985.

[6] Y. L. C. de Jong, "A CDMA Based Bidirectional Communication System for CATV Networks," Nederlands Elektronica- en Radiogenootschap, nr. 4, 1996, pp. 159-165.

[7] S. Lin and D. J. Costello, *Error Control Coding: Fundamentals and Applications*, Prentice Hall, 1983.

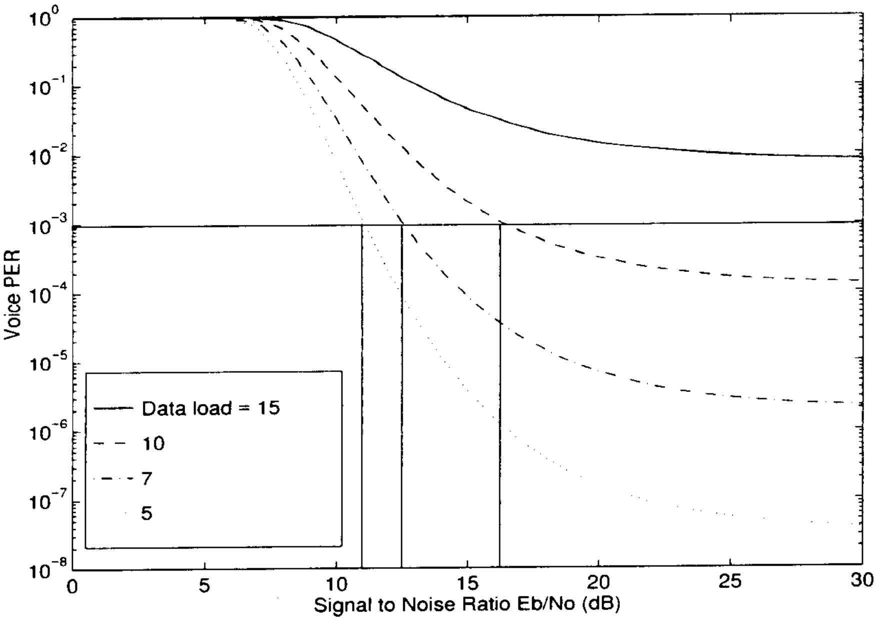

Fig. 1. Voice PER versus E_b/N_0 with voice traffic load = 11 and various data traffic load.

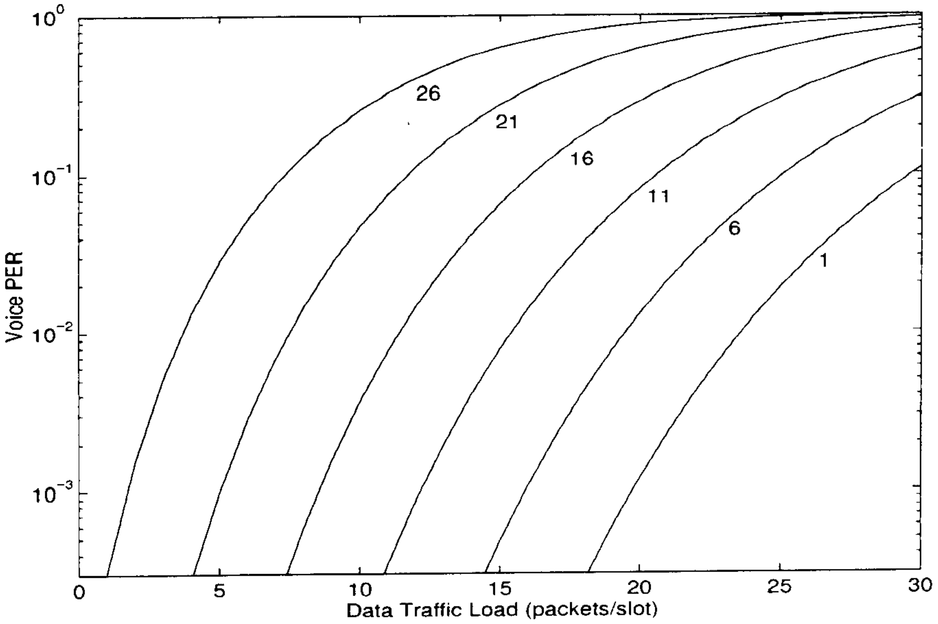

Fig. 2. Voice PER versus data traffic load with various voice traffic loads.

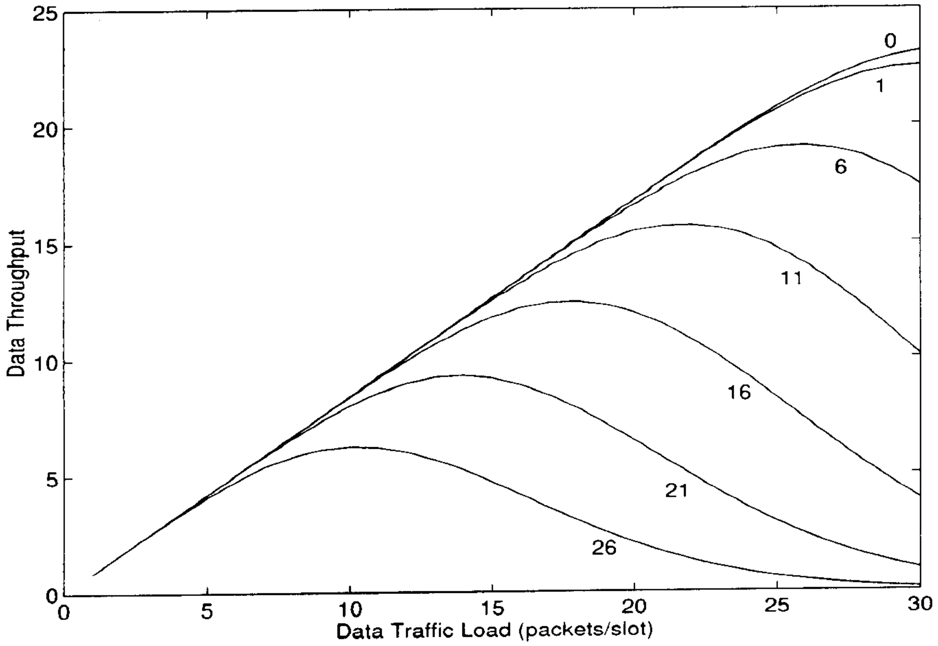

Fig. 3. Data throughput (packets/slot) as a function of the offered data traffic load with various voice traffic loads. Data code rate = 0.84.

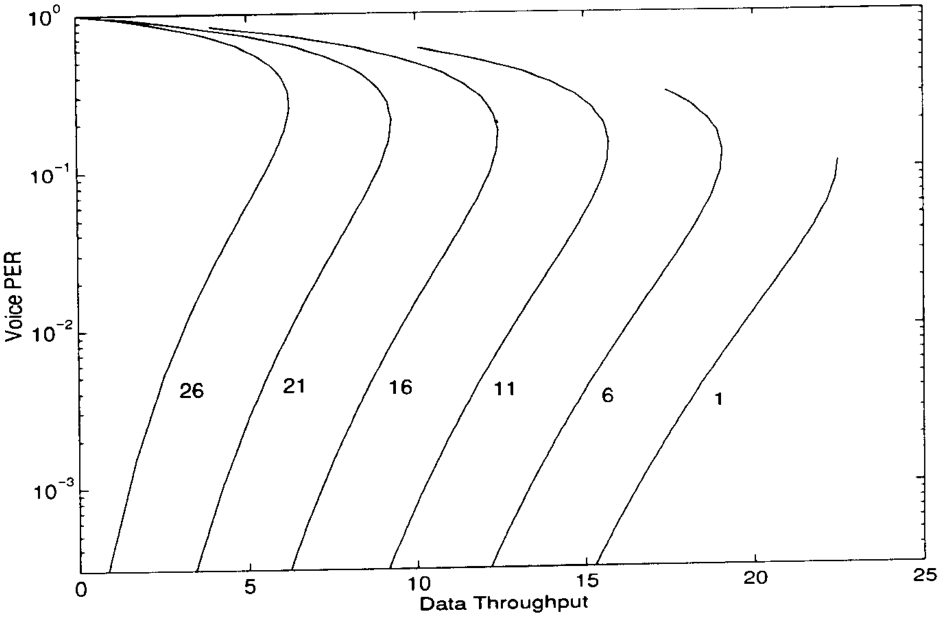

Fig. 4. Tradeoff between voice PER and data throughput with various voice traffic loads. Data code rate = 0.84.

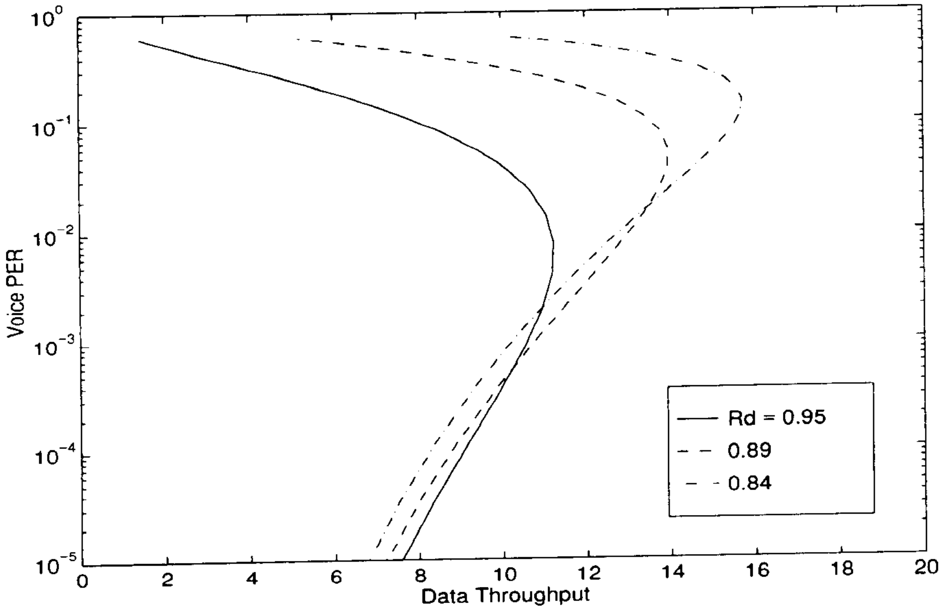

Fig. 5. Effect of data code rate on the tradeoff between voice PER and data throughput. The voice traffic load is 11. For voice PER between 10^{-2} and 10^{-3}, data code rate = 0.89 gives higher throughput.

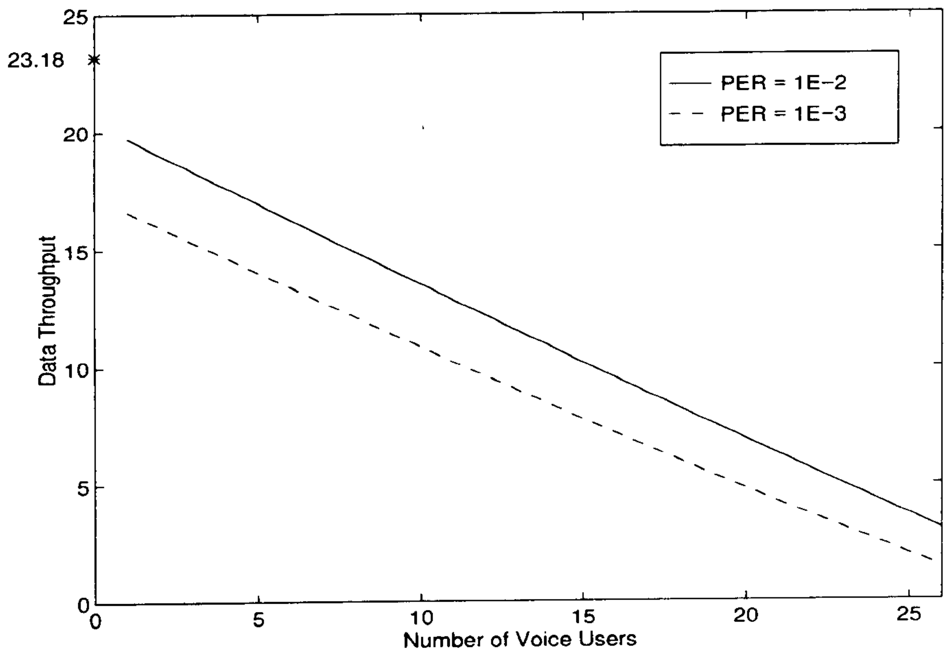

Fig. 6. Maximum supportable data throughput as a function of voice traffic load which require a specified PER. Data code rate = 0.84.

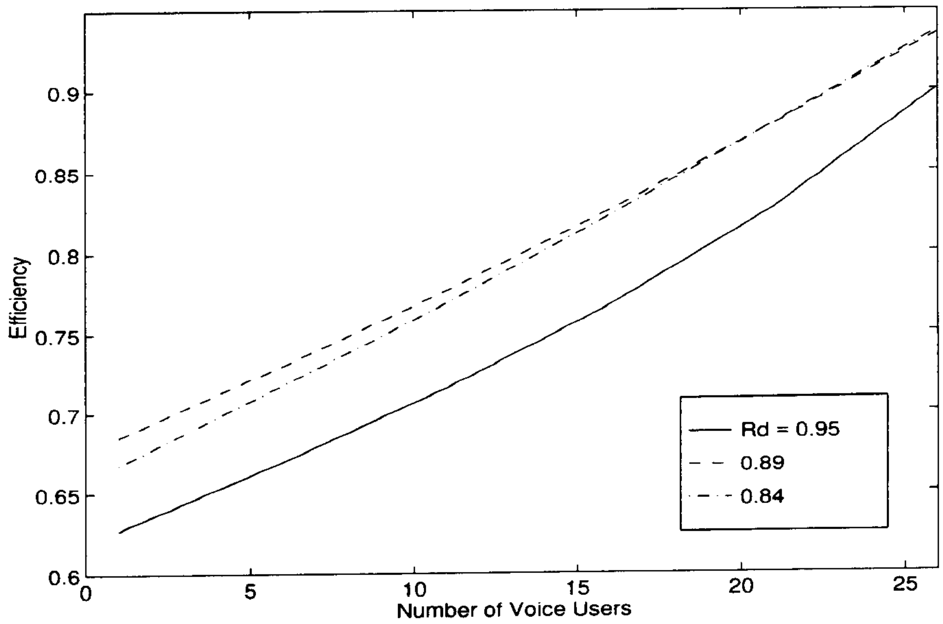

Fig. 7. The channel efficiency as a function of voice traffic load with different data code rates. Voice PER is required to be no higher than 10^{-2}.

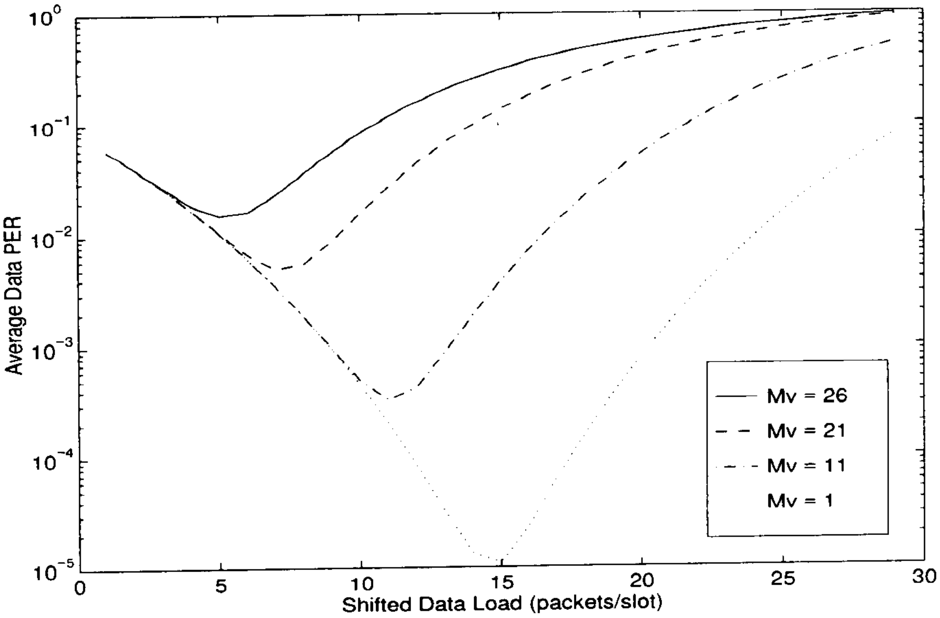

Fig. 8. Average data PER as a function of the data traffic load allocated in the integrated traffic channel. The total data traffic load is 30 packets/slot.

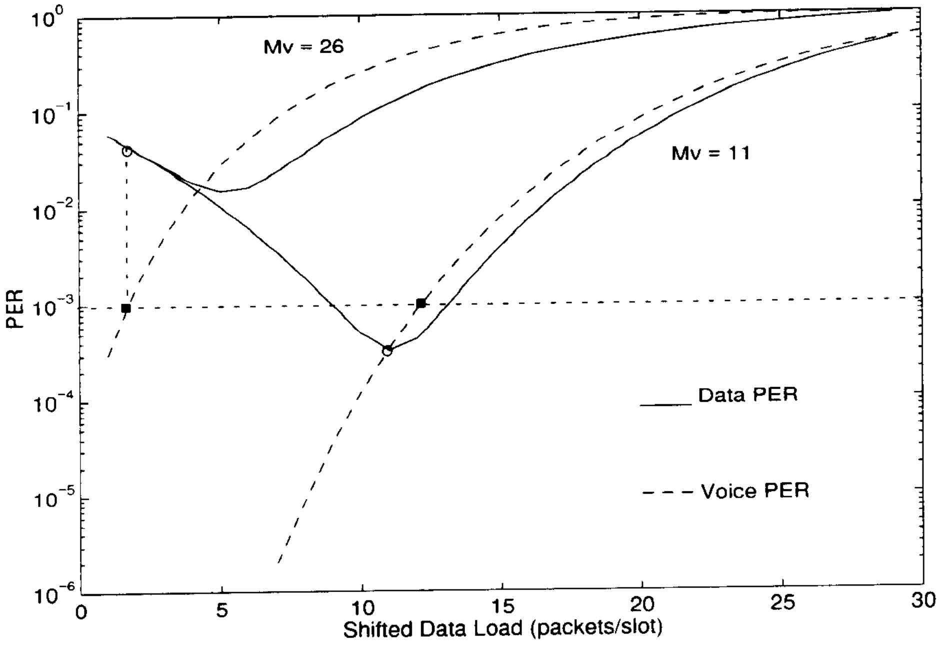

Fig. 9. Tradeoff between data PER and voice PER as a function of the data traffic load allocated in the integrated traffic channel.