CBR Packetized Voice transmission in IEEE802.11 Networks

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Abstract

The IEEE802.11 standard for wireless local area networks allows the coexistence of asynchronous and time-bounded transmissions using the DCF and PCF modes of operation. In this paper, we present the integration of packetized voice and data traffic over an IEEE802.11 BSS network and we analyze its performance in terms of maximum number of supported conversations and minimum bandwidth available for data transfers. The use of echo cancellation is considered and its effect on network performance is also analyzed.

1. Introduction

The IEEE802.11 standard specifies the coexistence of Distributed Coordination Function (DCF) and Point Coordination Function (PCF) in the MAC sublayer architecture [1]. DCF was developed for asynchronous data transmission, where all the stations share the medium using the CSMA/CA protocol and a random backoff mechanism, while PCF was developed for supporting time-bounded services, where a point coordinator (the Access Point of the Basic Service Set - BSS) determines which station has the right to transmit.

The transfer of real-time traffic, like voice, over packet networks is rapidly gaining acceptance, although many doubts have been arisen concerning the ability of IEEE802.11 networks to support voice services. At present, little work has been done to model the performance of the IEEE802.11 protocol in case of real-time transmissions. Visser [2] simulates the combination of speech traffic and data traffic over an IEEE802.11 network using statistical multiplexing and assuming that the voice activity occurs between stations in different BSSs. He concludes that the number of possible voice conversations is low and the performance is poor. In [3] and [4], Crow's simulations suggest that an echo canceller is required for handling on/off speech traffic exchanged among different BSSs. Romans [5] presents a hybrid protocol for wireless LANs which combines both TDMA access mechanism to support voice and CSMA/CA access mechanism to support data. This hybrid protocol is designed for use on a frequency hopping system and offers up to 4 reliable voice connections. At [6] a modified DCF access mechanism is proposed in order to provide real-time applications.

In this paper, we examine the characteristics of the service that voice traffic experiences when it is supported by the PCF access method of an IEEE802.11 LAN, while the DCF access method supports data traffic. We present a model of network performance that estimates an upper bound on the number of voice conversations that a BSS can handle, while keeping low voice packet delay and guaranteeing predetermined minimum bandwidth for data traffic. Our studies were performed using 64 kbps PCM without silence detection. Results are derived for scenarios with and without echo cancellation. We also assume an error free channel in order to focus on evaluating the PCF performance.

Section 2 describes the method that is used for integrating voice and data on an IEEE802.11 BSS. In Section 3, we present the analysis of a model that allows us to examine the performance achieved for voice traffic support and its effect on the bandwidth available to data stations. Finally, Section 4 presents extensive numerical results.

2. System Description

We consider a BSS network that can employ the combination of PCF and DCF functions. That network uses:

- a Point Coordinator (PC), that is the Access Point of the BSS,
- data only stations that use DCF to access the medium and to communicate with the PC and all other stations, and
- voice stations that support also data and use DCF to establish connections through the PC. These stations use PCF for packetised voice transmission.

During the network operation, both functions are supported using a time-sharing mechanism. The period of time the DCF operates is called the Contention Period

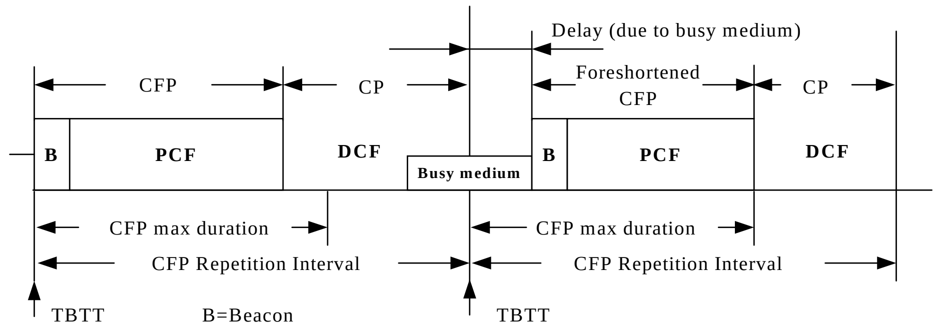

Figure 1. The CFP/CP alternation

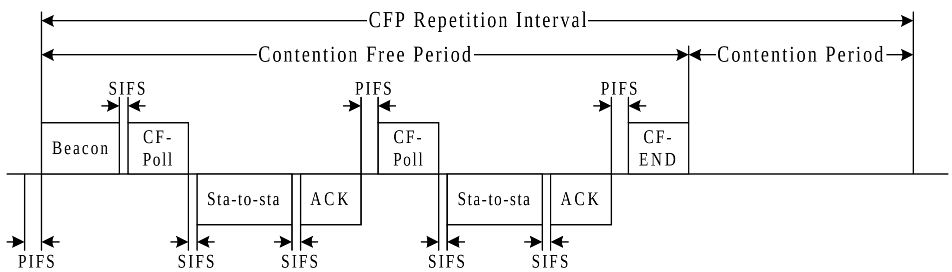

Figure 2. Voice transmission over an IEEE802.11 BSS network

(CP), implying that the stations contend for access, while the period of time the PCF operates is called the Contention Free Period (CFP), implying that the stations do not contend for access. The CFP always starts on a predefined time instance, which is determined by the PC and is called Contention Free Repetition Interval (CFPRI). The initiation of CFP is signed by a beacon frame transmission. The alternation of the CFP and CP periods is shown in Figure 1.

The PC controls the length of the CFP interval based on the available traffic and the size of its polling list. The PC may terminate any CFP by sending a CF-END frame at or before a maximum duration, called *CFPmaxDuration*. If at the nominal beacon transmission time, called Target Beacon Transmission Time (TBTT), the medium is busy due to DCF traffic, then the beacon is delayed and the CFP is foreshortened by the amount of this delay. In this case, the PC ends the CFP no later than TBTT plus the value of *CFPmaxDuration*. The amount of time that the beacon is delayed has a maximum value. Additionally, since the actual duration of CFP and CP may vary, the selection of *CFPmaxDuration* must allow a minimum duration of CP at which at least one data frame can be sent. The limitation of data traffic to a minimum bandwidth decreases the throughput of DCF and increases the data transmission delay depending on the offered load. All the time parameters that define the coexistence of the PCF and DCF are contained in the Beacon frame and the stations of the BSS are informed accordingly by receiving this frame.

The PCF is used for voice traffic as follows: Each voice station desiring to make a voice call issues a request that is placed on the polling list of PC. When the CFP starts, the PC sends a CF-Poll to the first station in the polling list. This station sends its voice packet to the other station in the BSS, no later than SIFS time after receiving the CF-Poll from the PC. When the destination station receives the voice packet, a DCF ACK frame is returned to the source station and the PC waits a PIFS interval following the ACK frame, before polling the next station in the polling list. Figure 2 depicts the rules under which voice packets are transmitted during CFP.

In this work we assume that voice stations generate traffic at 64 kbps constant rate. We consider that a voice packet is generated every CFPR interval. So the voice packet size depends on the duration of the CFPR interval. A voice packet is transmitted over the network each time the station is being polled by the PC. If a new packet is generated before an old packet has been transmitted, the old packet is discarded. In order to provide voice quality comparable to the telephone network, we consider that all stations on the polling list are polled once during each CFP. That procedure limits the probability of lost packets, since the time between two successive polling instants of the same station is close to the voice packet generation

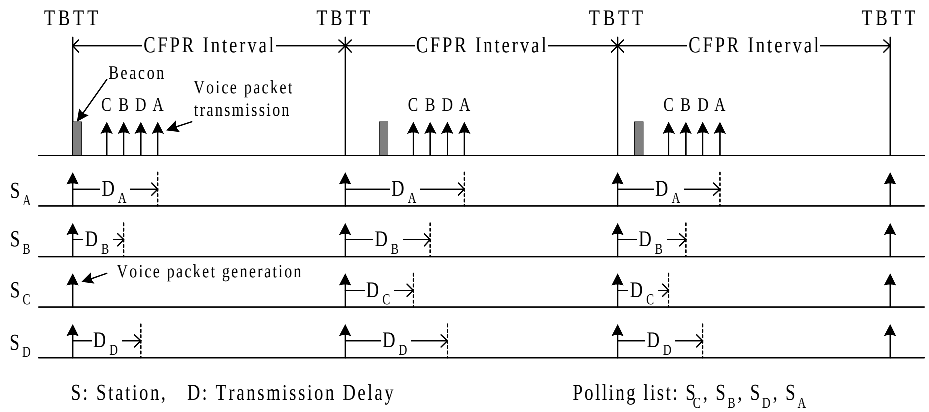

Figure 3. Voice traffic management

interval. This time is not exactly equal to the packet generation interval due to the fact that the CFP is sometimes foreshortened. This causes degradation of voice quality since some packets may be discarded. For solving this problem, we assume that each voice station starts sampling at a TBTT instant and so a new packet is generated every TBTT instant, as shown in Figure 3. In this case the voice packet of a station suffers a variable delay until the polling instant of that station arrives but the packet is not lost. All stations are synchronized using the Timing Synchronization Function (TSF) [1].

3. System Performance Analysis

Let T_{CFPR}, T_{CFP} and T_{CP} denote the CFPR, CFP and CP intervals respectively. Then, according to Figure 1:

$$T_{CFPR} = T_{CFP} + T_{CP} \tag{1}$$

The values of T_{CFP} and T_{CP} may vary but their sum is always constant. For calculating the maximum number of conversations that can be accommodated by PCF, data stations are provided with a minimum bandwidth, which is defined by the IEEE802.11 standard as:

$$T_{\min CP} = T_{\max MPDU} + 2SIFS + 2a + 8T_{ACK} + DIFS$$

where a is the parameter *SlotTime* that a data station uses as time unit for updating its backoff counter, while $T_{\max MPDU}$ and T_{ACK} are the transmission durations of a maximum length data frame and an ACK frame respectively. If $\max Payload$ is the size of the maximum data frame, H_{PH} is the physical layer header, H_{MAC} is the MAC header, ACK is the size of the ACK frame including the physical and the MAC headers and R_C is the channel bit rate, then:

$$T_{\max MPDU} = \frac{H_{PH} + H_{MAC} + \max Payload}{R_C} \text{ and } T_{ACK} = \frac{ACK}{R_C}$$

The CFP shall be maximum, when the CP is minimum since their sum is constant, but as we mentioned before, the CFP may be foreshortened due to the DCF traffic. Let T_{FS} be a random variable that describes the delay at the start of the CFP. According to [1] the maximum value of this random variable is:

$$T_{\max FS} = T_{RTS} + T_{CTS} + T_{\max MPDU} + T_{ACK} + 3SIFS$$

where $T_{RTS} = \dfrac{RTS}{R_C}$ and $T_{CTS} = \dfrac{CTS}{R_C}$

with RTS and CTS the sizes of RTS and CTS frames including physical and MAC headers. Whenever the CFP is foreshortened, some stations may need to discard their voice packets. Therefore, the upper bound of conversations for the maximum CFP is found for the time length $T'_{CFP} = T_{\max CFP} - T_{\max FS}$ and equation (1) is modified as:

$$T_{CFPR} = T_{\max FS} + T'_{CFP} + T_{\min CP} \tag{2}$$

According to Figure 2, the time length T_{Con} of a connection between stations for exchanging voice packets during PCF is given by:

$$\begin{aligned} T_{Con} &= 2\left(T_{CF-Poll} + SIFS + T_{vp} + SIFS + T_{ACK} + PIFS\right) \\ &= 2\left(T_{CF-Poll} + T_{vp} + T_{ACK} + 2SIFS + PIFS\right) \end{aligned} \tag{3}$$

where $T_{CF-Poll}$ is the transmission time of a CF-Poll frame with size $CF-Poll$ bits (including the physical and the MAC layer headers), T_{vp} is the transmission time of a voice packet with size $T_S R_S$ bits, T_S is the voice packet generation interval and R_S is the voice sampling rate. In our system we take $T_S = T_{CFPR}$. So

$$T_{CF-Poll} = \frac{CF-Poll}{R_C} \text{ and } T_{vp} = \frac{H_{PH} + H_{MAC} + T_S R_S}{R_C}$$

Let $N_{\max}$ denote the maximum number of voice conversations that can be handled during PCF, then

$$T'_{CFP} = PIFS + T_{Beacon} + N_{\max} T_{Con} + T_{CF-END} \quad (4)$$

where

$$T_{Beacon} = \frac{Beacon}{R_C} \text{ and } T_{CF-END} = \frac{CF-END}{R_C}$$

Using (2) and (4) we can calculate the maximum number of conversations (and the maximum number of voice stations: $2N_{\max}$) an IEEE802.11 BSS network can support for various values of CFPR interval, which is:

$$N_{\max} = \big(T_{CFPR} - T_{\max FS} - PIFS - T_{Beacon} - SIFS - T_{CF-END} - T_{\min CP}\big) / T_{Con} \quad (5)$$

Since the MAC frame has a maximum size ($\max Payload$), the size of the voice packet must be up to this value. That causes the duration T_S of a voice packet generation interval and its equal CFPR interval T_{CFPR} to have an upper bound:

$$T_{\max S} = T_{\max CFPR} = \frac{\max Payload}{R_S}$$

which is equal to 289 msec for 64 kbps CBR packetized voice in an IEEE802.11 WLAN. From Figure 3 it is obvious that in our system the delay of the voice packets is shorter than the CFPR interval and in case of the maximum CFPR interval, the delay is shorter than 289 msec. Quality of service (QoS) parameters for voice typically limit maximum delay to 25 msec without echo cancellation and 500 msec using echo cancellation. When echo cancellation is used, the above analysis satisfies the maximum delay requirement, but without using echo cancellation, the analysis has to be modified.

In this case we must deal with the voice packet delay of the last station on the polling list, since this station has the greatest delay. This delay must be constrained to 25 msec and the maximum number of conversations is defined by this delay. Defining that $D_{\max LS}$ is the maximum voice packet delay the last station on the polling list can suffer with echo cancellation, $D'_{\max LS}$ is the maximum packet delay the last station on the polling list can suffer without echo cancellation, $d_{\max}$ is the bound of 25 msec and $N'_{\max}$ is the maximum number of conversations that correspond to the no echo cancellation case, then we have:

$$\begin{aligned} D_{\max LS} &= T_{\max FS} + T_{Beacon} + SIFS + N_{\max} T_{Con} \\ D'_{\max LS} &= T_{\max FS} + T_{Beacon} + SIFS + N'_{\max} T_{Con} \end{aligned} \quad (6)$$

$D_{\max LS} < 500$ ms for all values of $N_{\max}$ that derive for various CFPR intervals, but it does not occur the same for $D'_{\max LS}$. We can calculate the $N'_{\max}$ by taking into account the requirement $D'_{\max LS} \le d_{\max}$ and (6)

$$N'_{\max} = \begin{cases} N_{\max}, & \text{if } D'_{\max LS} \le d_{\max} \\ \dfrac{d_{\max} - T_{\max FS} - T_{Beacon} - SIFS}{T_{Con}}, & \text{otherwise} \end{cases} \quad (7)$$

From (7) we can find the maximum number of conversations for different CFPR intervals, without echo cancellation and for supporting the previously defined QoS requirements. Another approach is to calculate the probability the voice packet delay is greater than $d_{\max}$. The voice packets that present greater delay than $d_{\max}$ are discarded and the voice quality deteriorates, unless the above probability is very small. More specifically, if D_{LS} is the random variable describing the voice packet delay of the last station on the polling list, S is the random variable that describes the delay due to the DCF traffic and N is the number of conversations, then:

$$D_L = S + T_{Beacon} + SIFS + NT_{Con}$$

where $C = T_{Beacon} + SIFS + NT_{Con}$ is constant. So, if the Probability Distribution Function (PDF) of the random variable S is known, we can find the complementary probability $P[D_{LS} > d_{\max}]$, that the voice packet delay is greater than $d_{\max}$:

$$\begin{aligned} P[D_{LS} > d] &= 1 - P[D_{LS} \le d] = 1 - F_{D_{LS}}(d) \\ &= 1 - F_S(d - C) \end{aligned} \quad (8)$$

where $F_{D_{LS}}(d)$ and $F_S(s)$ are the PDFs of the random variables D_{LS} and S respectively. For example, if the delay S follows the exponential distribution then:

$$P[D_{LS} > d] = e^{-\lambda(d-C)}, \quad d \ge 0$$

considering that $\lambda = 5/T_{\max FS}$.

The upper bound of channel utilization CU_v for voice transmissions, is given by:

$$CU_{\max v}\% = \frac{2N_{\max}\frac{T_S R_S}{R_C}}{T_{CFPR}} 100 = \frac{2N_{\max} R_S}{R_C} 100$$

for cases with echo cancellation (9)

$$CU'_{\max v}\% = \frac{2N'_{\max}\frac{T_S R_S}{R_C}}{T_{CFPR}} 100 = \frac{2N'_{\max} R_S}{R_C} 100$$

for cases without echo cancellation

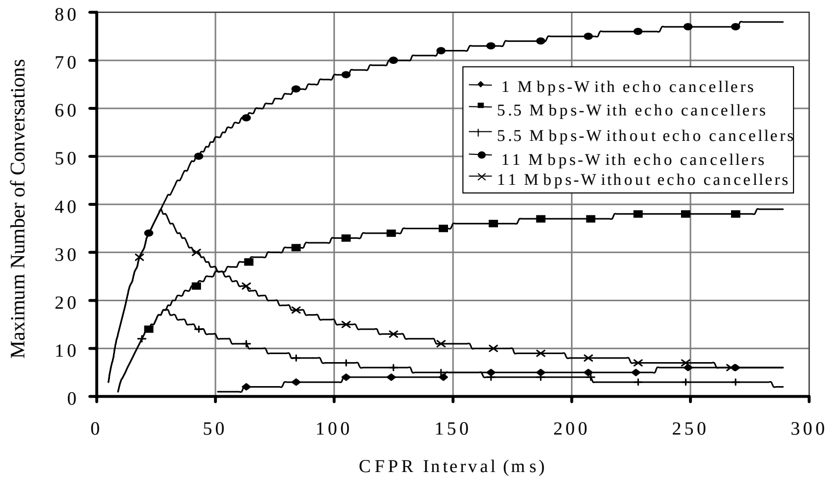

Figure 4. Maximum number of conversations with and without echo cancellation

while the remaining bandwidth BW_d for asynchronous data transmissions is given by:

$$\begin{aligned} BW_d\% = (&T_{CFPR} - T_{\max FS} - PIFS - T_{Beacon} - SIFS \\ &- N_{\max} T_{Con} - T_{CF-END})100/T_{CFPR} \\ &\text{for cases with echo cancellation and} \\ BW_d'\% = (&T_{CFPR} - T_{\max FS} - PIFS - T_{Beacon} - SIFS \\ &- N_{\max}' T_{Con} - T_{CF-END})100/T_{CFPR} \\ &\text{for cases without echo cancellation.} \end{aligned} \tag{10}$$

4. Numerical results

In this section, we present and discuss numerical results showing the performance of the IEEE802.11 BSS network under voice and data traffic. The results of our analysis are derived considering that: MAC header = 34x8 bits (including the FCS field), Physical header = 16x8 bits, ACK = 30x8 bits, RTS = 36x8 bits, CTS = 30x8 bits, CF-Poll = 50x8 bits, CF-END = 36x8 bits, Beacon = 106x8 bits, SIFS = 10 usec, PIFS = 20 usec, DIFS = 50 usec, *SlotTime* = 20 usec, $T_{CFPR} = T_S = 1, 2, ...289$ msec and Channel Bit Rate = 1, 5.5 and 11 Mbps.

Figure 4 shows the dependence of the maximum number of conversations, which can be handled by our system, to the CFPR interval for various channel bit rates. In case of using echo cancellation, we notice that the number of conversations increases as the CFPR interval and the channel bit rate increase, since fewer overhead is used per information block unit and the bandwidth allocated for voice transfers becomes larger. Further, the performance of PCF for 1 Mbps channel bit rate is poor (low number of conversations) and the voice transfers are feasible if the CFPR interval is greater than 51 ms. Without echo cancellation, there is a specific value of the CFPR interval at which the PCF supports the maximum number of conversations and the delay of voice packets is limited to 25 ms. Beyond that value, the number of conversations decreases, while the CFPR interval increases. The reason is that we limit the CFP at a specific length in order to satisfy the delay constraint, but the voice packet size becomes bigger as the CFPR interval increases and so the PCF can handle fewer voice stations during the CFP. For 1 Mbps channel bit rate, the BSS network cannot provide voice traffic without echo cancellation.

Figure 5 illustrates the upper bound of the voice packet delay with and without echo cancellation. We observe that the voice packet delay remains lower than 500 msec with echo cancellation and lower than 25 msec without echo cancellation as the CFPR interval grows. The distribution of the delay of the DCF traffic affects the complementary probability distribution of the voice packet delay, as we can see in Figure 6, where the distribution of the delay of the DCF traffic is exponential, the channel bit rate is 1 Mbps and the CFPR interval is 51 ms. Great interest presents the probability the voice packet delay is greater than 25 ms, since it indicates the probability of the number of packets that don't satisfy the delay constrain and must be discarded when echo cancellation is not available. In our example this probability equals to 0.02.

According to Figure 7, the voice channel utilization upper bound (CFPR interval percentage) increases as the CFPR interval increases, when we use echo cancellation. This occurs because the enlargement of the CFPR interval allows more voice stations to be placed on the polling list of the PCF, while the data stations can use only the minimum available bandwidth for the DCF. Without echo

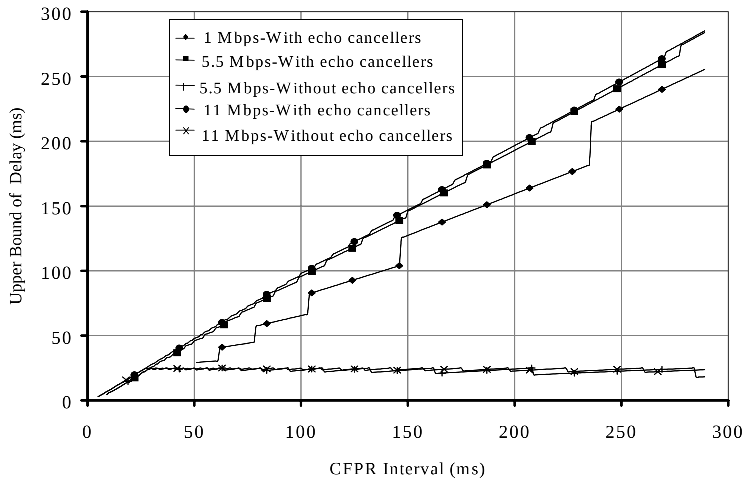

Figure 5. Upper bound of voice packet delay versus CFPR interval

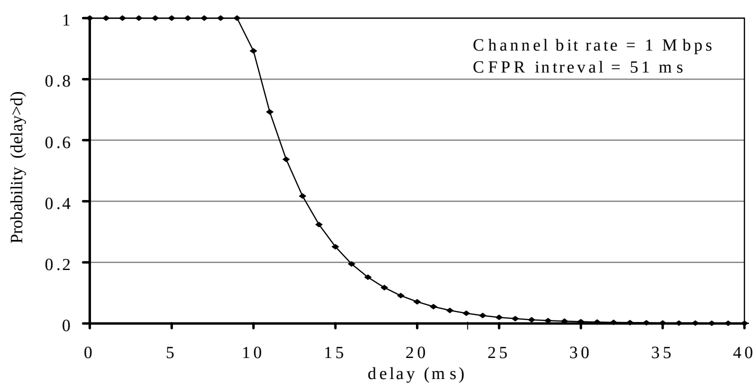

Figure 6. The complementary probability distribution of voice packet delay

cancellation, the maximum value of voice channel utilization is reached when the CFPR interval causes 25 msec voice packet delay. For greater values of the CFPR interval the voice channel utilization decreases, since the number of conversations becomes lower in order to provide voice quality comparable to telephone networks.

Finally, Figure 8 depicts the percentage of bandwidth of the CFPR interval that remains for data transmissions. With echo cancellation the available data bandwidth is limited to the minimum value defined by [1] and so its portion decreases as the CFPR interval increases. On the other hand, without echo cancellation, the portion of bandwidth that is devoted to data transmission becomes lower as the CFPR interval increases, until the voice packet transmissions approach the delay constraint. After that point, the available data bandwidth increases, since the number of conversations decreases.

5. Conclusions

In this paper, we described and analyzed the integration of data with constant bit rate packetized voice over an IEEE802.11 BSS network. According to the results of our analytical approach, the performance of such a system is low for voice transfers with channel bit rate equal to 1 Mbps and in this case echo cancellation is essential. For higher transmission rates, more conversations can be accomplished as the CFPR interval increases. A larger number of conversations is feasible when echo cancellation is used but the bandwidth for data transmissions remains minimum. On the other hand, the available bandwidth for data transfers increases in the case of networks without echo cancellation, since the number of conversations becomes smaller when the CFPR interval increases above a specific value in order to support the required voice quality.

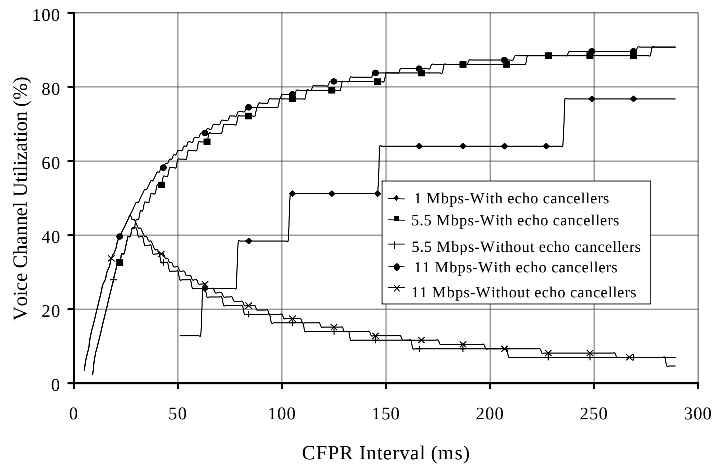

Figure 7. Upper bound of voice channel utilization versus CFPR interval

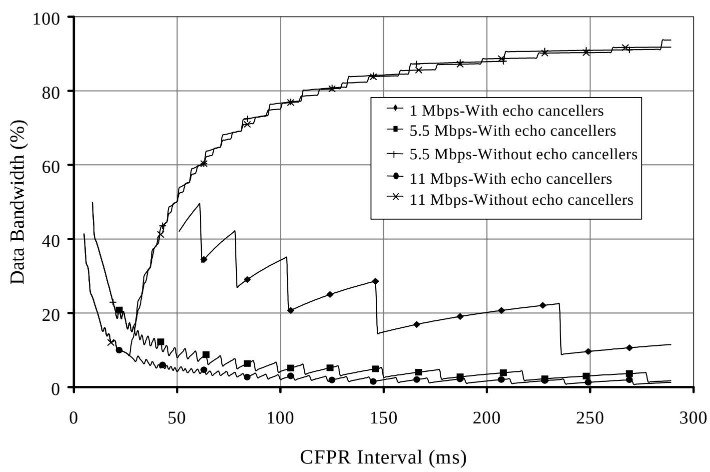

Figure 8. Available bandwidth for data transmissions versus CFPR interval

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