Resource Planning for Voice over Wireless Both-Way Transmission Media

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Abstract. *The medium in IEEE 802.11- and IEEE 802.16-based networks for voice communications can be considered "both-way" - for transmission and reception. Therefore, the packet arrivals for voice dialogue services in such networks are not strictly independent. In this paper, we discuss the traffic capacity in the call (network) layer and suggest accounting for the impact of the correlated nature of two-way voice conversations on performance estimation. We present analytical results and numerical examples.*

Keywords

IEEE 802.11, IEEE 802.16, voice call capacity, WLAN, WMAN.

1. Introduction

There has been an increased interest in using IEEE 802.11- and IEEE 802.16-based networks (called also WLAN (WiFi) and WMAN (WiMAX), respectively) for voice and other real-time dialogue services. Three common features of these networks (together with wired LAN) are that they are almost always end–user networks, that all of them need a medium access control (MAC) layer, and that they have both-way transmission media.

The use of voice over WLAN/WMAN is curtailed by poor voice performance capability. Although the proclaimed PHY data rate for WLAN/WMAN is up to several tens of Mbps and the voice codec output rate is not more than 64 kbps, in practice a WLAN/WMAN network can support at most several tens of simultaneous calls [1]-[3].

Many solutions have been proposed for improving the voice capacity of IEEE 802.11/802.16 networks. These solutions explore features of VoIP traffic, and include:

A. Codec Bit Rate Reduction: Use of lower bit rate codec like ITU-T G723 and G729 or GSM 6.10 [4].

B. Header Compression: The small VoIP packets carry large headers. The voice packet payload is 10 – 30 bytes, while the RTP/UDP/IP/MAC protocol overheads are as large as double or triple the payload, not to mention the PHY preamble and the control packets. As a result, in some cases the overall efficiency has been measured to be less than 3% [3]. Different header compression techniques have recently been proposed (see [3]–[5] and references therein).

C. Control Overhead Reduction: The so-called enhancements IEEE 802.11e and IEEE 802.16e of the legacy standards are oriented to real-time traffic applications, and they propose VoIP packet control mechanisms that reduce the control overhead. Various other control overhead reduction techniques have also been proposed in the literature [6], [7].

D. Silence Suppression: Enhanced voice codecs such as ITU-T G723.1A and G 729B use silence suppression to prevent voice packet transmission during inactivity [6]-[8]. Inactivity periods occupy up to 60% of the total call duration, and elimination of wasted bandwidth can double the voice capacity [3].

E. Accounting for Dialogue Packet Correlation and Both-Way Media: During a conversation, there is a correlation between party talk spurts. When one party is active (ON) and generating packets, the other party is usually inactive (OFF) and not generating packets. With fixed and mobile networks where the transmission and reception paths are separated, the above correlation does not matter. By contrast, LAN, WLAN and WMAN have one common medium for both ways of transmission (an exception is WiMAX realized with frequency division duplexing). One can expect dialogue correlation to reduce the total traffic peakness, and therefore to reduce resources needed to serve the traffic.

The goal of this paper is to evaluate the impact on performance (if any) of the correlated speech packet bursts of voice conversations over both-way transmission media. To the best of our knowledge, the present article is the first to address such an evaluation. We begin by describing a generalized model for voice call performance that takes into account the dialogue correlation. We then present examples and numerical results.

2. The Voice Call Capacity Traffic Model

The packet form of voice transmission differs considerably from other data transmissions. The VoIP traffic is streaming with stringent delay requirement and prolonged activity periods of generating packets. During activity periods, the data transmission rate of encoded voice is relatively low. The packets are comparatively short with constant length.

Due to the great interest in using IEEE 802.11 and 802.16 standards for voice transmission, there are numerous publications on modeling the voice call performance. Examples include [1]-[3], [6]-[8], [9], and the references therein.

The models suggested in the literature so far take into consideration some or all of the above-mentioned voice traffic features, but do not account directly for the two-way correlated nature of voice conversations. Exceptions (for IEEE 802.11 only) are the simulations in [7], [8] and the analytical model in [10]. The latter model is not developed specifically for voice transmission, but only applied in that context. There are also interesting models for wireless networks obtained using a cross-layer framework [11]-[13]. Such models integrate information theory (the physical layer), scheduling presentation (the MAC layer) and queuing theory (the network layer).

Our purpose is to present a model that is effective for the evaluation of the correlated nature of voice conversations and applicable for any network like LAN, WLAN, and WMAN with both-way transmission medium. In particular, we consider the traffic capacity in the call (network) layer to assess the impact of the speech packet bursts correlation on the utilization of the bandwidth supplied by the lower layers.

2.1 The Model

VoIP packets experience two kinds of losses: *packet losses* and *talk spurt losses.* Packet losses are due to the fact that there is one channel for all packets, and when a packet is being transmitted, it is always possible for another one to arrive. These losses are easy to overcome by means of a relatively short buffer because of the low rate of voice transmission. The buffering that already exists in WLAN/WMAN provides a natural solution to the packet loss problem. Talk spurt losses occur when the number i of simultaneously active sources is bigger than the number of network resources (n). Dealing with talk spurt losses by means of a buffer is not appropriate - the buffer would have to be large enough because of the voice transmission's long active periods, and hence would introduce unacceptable delay. Therefore, we will use a bufferless talk spurt packet loss model similar to [14, pp. 141-145]. In practice, for network design purposes, the talk spurt losses in particular should be managed not by buffers, but by properly selecting the available medium transmission rate C or properly designed *connection admission control*.

2.2 Notations

C = total medium rate allocated to voice transmission;
c = voice source rate in ON state;
N = number of voice packet traffic sources;
n = number of transmission resource units;
P_i= probability that i traffic sources are in active state;
$P_{i>n}$= probability that the system is in a state with talk spurt losses;
P_{TSL}= probability that a packet is lost due to a talk spurt;
P_{TSLL}= probability that a packet is lost due to a talk spurt only for states $i > n$;
S = number of subscriber stations;
T_{on} = ON state mean duration;
T_{off} = OFF state mean duration;
α = activity factor.

The units of C and c (bits per second, packets per second, etc.) are not important, as long as they are the same.

2.3 Analytical Evaluation of Packet Losses

The available resources can be represented by the number of transmission resource units (not necessarily an integer value):

$$n = C/c\,. \tag{1}$$

The probability that a source is in active state (the *activity factor*) is given by:

$$\alpha = [T_{on}]/[T_{on} + T_{off}]\,. \tag{2}$$

The probability that exactly i out of N traffic sources are in an active state is:

$$P_i = \binom{N}{i} \alpha^i (1-\alpha)^{N-i}, \quad 0 \le i \le N. \tag{3}$$

Talk spurt losses are possible if $n < N$. The probability that the system is in a state with talk spurt losses is:

$$P_{i>n} = \sum_{i=\lceil n \rceil}^{N} P_i = \sum_{i=\lceil n \rceil}^{N} \binom{N}{i} \alpha^i (1-\alpha)^{N-i}\,, \tag{4}$$

where $\lceil n \rceil$ denotes the minimum integer value $\geq n$.

The probability that a packet is lost is given by the ratio of the lost packet rate to the arriving packet rate:

$$P_{TSL} = \frac{\sum_{i=\lceil n \rceil}^{N} (i-n)c \binom{N}{i} \alpha^i (1-\alpha)^{N-i}}{\sum_{i=1}^{N} ic \binom{N}{i} \alpha^i (1-\alpha)^{N-i}} =$$

$$= \left[\sum_{i=\lceil n \rceil}^{N} (i-n) \binom{N}{i} \alpha^i (1-\alpha)^{N-i} \right] \Big/ \left[N\alpha \right]. \tag{5}$$

2.4 Dialogue Talk Spurt Correlation and Both-Way Medium

In a network, a Subscriber Station (SS) communicates with parties that are behind the Base Station (BS) and the BS re-transmits the packets, so one can assume that the dialogue is between the SS and the BS. When there are S packet sources from SS transmitters, there are additionally S packet sources behind the BS. If there are no voice communications between SSs within the same network (which is a reasonable assumption for an access network), there are altogether $N = S + S$ packet sources.

Our main observation is that the ON/OFF pattern of a SS transmitter corresponds to an OFF/ON pattern of the party transmitter behind the BS. This reduces the competition among stations for the medium that can be evaluated with traffic dispersion reduction.

Based on this fact, we suggest a simple way to take into account the impact of the dependence between activity periods for transmission and reception. Specifically, we propose substituting N with $N/2$ and α with 2α. in (6). We illustrate the effect of this change with the following numerical experiments.

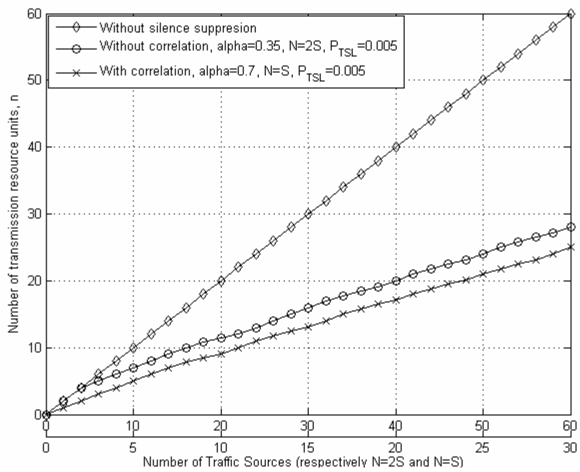

Fig.1. Network dimensioning with α=0.35 and PTSL=0.5%.

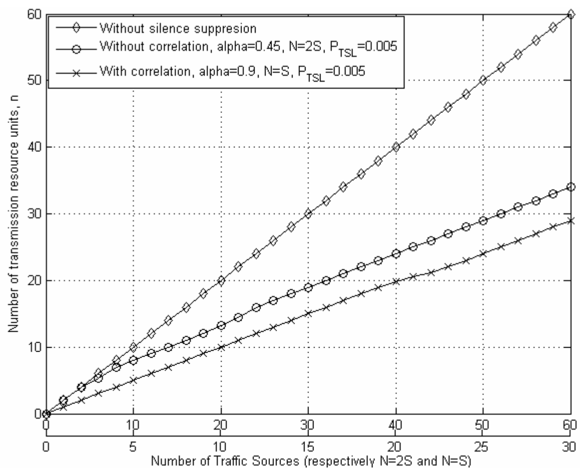

Fig.2. Network dimensioning with α=0.45 and P_{TSL}=0.5%.

Fig. 1 and Fig. 2 show the network dimensioning using equation (5) with typical values for the packet losses and activity factor [15]. The number of traffic sources would equal to the number of SSs when there is no dependence between transmission and reception activity, but, as explained before, it is in effect reduced by half when correlation is taken in consideration. We assume that each traffic source would require one transmission resource unit if there were no silence suppression. The actual total number of transmission units needed with silence suppression with and without taking into account the effect of correlation (rounded to the nearest integer) is plotted for two different values of α: 0.35 and 0.45. As expected, silence suppression decreases considerably the number of resource units needed to attain the target probability of packet losses. More importantly for our analysis, considering the voice talk spurts correlation over both-way media leads to additional reduction in the necessary bandwidth, and the difference is greater for larger values of α. The correlation effect appears significant enough to merit consideration in network design.

2.5 Packet Losses Evaluation only for States with Losses

Equation (5) gives average losses for all possible states, including states without losses. In state $i = N,$ the subscribers will suffer highest losses equal to $(N - n)/n$ on average, whereas in all states for which $i < n$, there are no losses.

Instead of P_{TSL} as computed from (5), we propose using a modified measure for the probability of packet losses, which takes into consideration the observation above and provides a more realistic loss calculation. The modified probability of losses, P_{TSLL}, is calculated as:

$$P_{TSLL} = \frac{\sum_{i=\lceil n \rceil}^{N} (i-n)c\binom{N}{i}\alpha^i(1-\alpha)^{N-i}}{\sum_{i=\lceil n \rceil}^{N} ic\binom{N}{i}\alpha^i(1-\alpha)^{N-i}} =$$

$$= \frac{\sum_{i=\lceil n \rceil}^{N} (i-n)\binom{N}{i}\left(\frac{\alpha}{1-\alpha}\right)^i}{\sum_{i=\lceil n \rceil}^{N} i\binom{N}{i}\left(\frac{\alpha}{1-\alpha}\right)^i} = \frac{\sum_{i=\lceil n \rceil}^{N} (i-n)\binom{N}{i}\left(\frac{T_{on}}{T_{off}}\right)^i}{\sum_{i=\lceil n \rceil}^{N} i\binom{N}{i}\left(\frac{T_{on}}{T_{off}}\right)^i}. \quad (6)$$

It is easy to derive the relationship between P_{TSLL} and P_{TSL}:

$$\frac{P_{TSLL}}{P_{TSL}} = \left[\sum_{i=1}^{N} i\binom{N}{i}\left(\frac{T_{on}}{T_{off}}\right)\right] \Big/ \left[\sum_{i=\lceil n \rceil}^{N} i\binom{N}{i}\left(\frac{T_{on}}{T_{off}}\right)\right]. \quad (7)$$

Fig. 3 shows network dimensioning obtained from equation (6) (rather than equation (5)) using P_{TSLL} as the norm for losses (we assume P_{TSLL} = 3%). The selection of 3% as the value for P_{TSLL} is arbitrary, and is only meant to provide an example. The assessment of voice degradation due to packet losses as measured by P_{TSLL} could be a subject of separate investigation as has already been done for P_{TSL} (see, for example, the excellent paper [15] and the references therein).

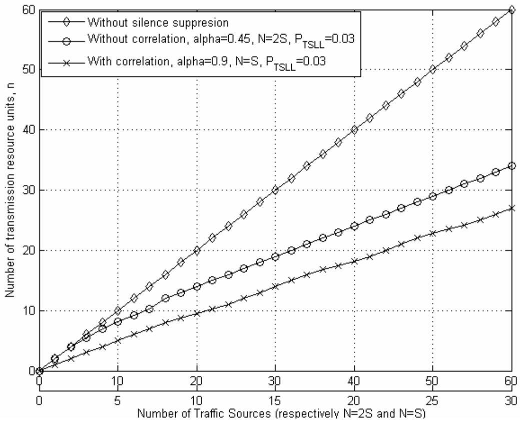

Fig. 3. Network dimensioning with α=0.45 and P_{TSLL}=3%.

3. Conclusions

In this paper, we discussed the influence of the voice talk spurt correlation over both-way media. The results obtained can be used for IEEE 802.11 and 802.16 packet voice network design and Call Admission Control management. The numerical results plotted in Fig. 1 – Fig. 3 show that taking into account talk spurt correlation can free up resources during the network planning process.

The model could be extended by considering additional details, such as the slow rate transmission during OFF periods, or the short periods with double talk when the silence party is interrupting the talking party.

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