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VOICE TRAFFIC PERFORMANCE OVER WIRELESS LINKS

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Key words: *Call admission control, Performance analysis, Quality of service, Voice activity detection, Voice over IP, Wireless access networks, Wireless channel*

Abstract: *Wireless resources dimensioning is a crucial task, which requires development of accurate resource estimation models. Providing quality of service (QoS) for Voice over IP traffic transmissions over wireless links requires an appropriate call admission control (CAC) algorithm development. This article aims at presenting an approach for call admission control mechanism dimensioning applicable to wireless access networks. The admission decision policy is based on the overall packet loss probability evaluation with respect to the wireless link dynamics and the scarce resources available at the wireless access point.*

INTRODUCTION

Despite emerging variety of IP-based multimedia services, Voice over IP (VoIP) services are still attractive to both end-customers and network operators. In contrast to the circuit-switched technologies, the challenge to the packet-based networks is to provide QoS guarantee through proper resource management in both core and access domains. Since wireless access technologies are inseparable part of our daily life, there exists a substantial interest in investigating their voice traffic performance. The major design issue is how to manage the scarce and expensive wireless resources, such that an acceptable QoS to be achieved. VoIP traffic transmission over wireless links encounters several issues and particularities, such as stringent norms of admissible delay; radio resource waste due to relatively long packet header, which could considerably exceed the packet payload, carrying voice frames; and time-varying channel conditions. In order to prevent wasting available resources a variety of voice codecs with voice activity detection (VAD) are developed.

CAC techniques in contemporary wireless networks have been a subject of intensive study [1], [2]. The vast majority of research on packetized voice traffic performance over wireless networks is carried out by means of simulation analysis. Since network service providers are mainly interested in system operation under real traffic load, QoS parameters, such as call blocking, packet losses, etc., are to be modeled as rare events. Unfortunately, this is a time and resource consuming process. An efficient alternative is to employ analytical methods for system performance evaluation.

A common assumption for model complexity reduction is data transmission in an error-free environment [3], [4]. In some proposals non-ideal channel conditions are taken into consideration, investigating the wireless channel influence on the system performance by means of extensive simulation models and tools [5].

The objective of this article is to propose a method of analytical investigation of an admission control scheme for streaming traffic flows by incorporating the time-varying nature of wireless transmission links when evaluating packet loss rates for voice applications.

SYSTEM DESCRIPTION

The system under investigation comprises *N* identical sources, each operating independently, and sending voice packets of constant rate during the active state only. A VoIP traffic source with VAD is considered and thus, represented as an ON-OFF traffic model. The state durations are assumed to be exponentially distributed. Independently of the technology used, an access point (AP) is dimensioned with a particular service capacity and aggregates the traffic of multiple VoIP sources over a wireless interface.

The basic data unit is a packet, which may contain multiple voice frames generated by a voice codec of particular type and supporting VAD. The wireless channel is assumed to be frequency flat and remains invariant per packet transmission. For such channels, the channel quality could be sufficiently described by the received signal-to-noise ratio (SNR). For this reason, the Nakagami-*m* fading model is adopted.

PERFORMANCE EVALUATION

The fundamental limitation of wireless communication systems is imposed by the time-varying transmission medium. The transmitted radio frequency signal propagates through different routes, experiencing scattering, diffractions and reflections, before it arrives at the receiver. In such a fading channel considerable SNR fluctuations could be experienced by the receiver. These can be statistically described by the Gamma probability density function of the instantaneous SNR γ per transmitted voice packet

$$(1) \qquad p_{ch}(\gamma) = \frac{m^m \cdot \gamma^{m-1}}{\bar{\gamma}^m \cdot \Gamma(m)} \cdot \exp\left(-\frac{m \cdot \gamma}{\bar{\gamma}}\right),$$

where $\bar{\gamma}$ is the average received SNR, $\Gamma(m)$ is the Gamma function, and m is the Nakagami fading parameter ($m \geq 1/2$). The channel model can be applied to a large class of fading channels (e.g., $m = 1$ corresponds to a Rayleigh channel). It also allows Ricean channels to be well approximated by Nakagami-*m* channels by one-to-one mapping between the Ricean factor *K* and the Nakagami fading parameter *m* [6].

Since the wireless medium is non-ideal, data could be corrupted at the receiver. It is quantitatively expressed by the average packet error rate (PER). Following the assumption of independent packet loss probability (PPL) and transmission error probability, the overall packet loss probability $P_{loss}^{(\gamma)}$ for given γ is defined as

$$(2) \qquad P_{loss}^{(\gamma)} = 1 - (1 - PPL) \cdot (1 - PER).$$

The packet loss probability (PPL) is governed by the insufficient amount of resources available at the AP to serve active voice connections. This is a case when the total input rate from active sources exceeds the AP's service rate. The design criteria aims at properly dimensioning service capacity in order to meet QoS requirements. The stringent requirements on voice traffic delay limit the methods of handling traffic losses. The packet-scale losses can be limited by operating a small-size buffer. Preventing talk-spurt losses by means of buffering is not feasible since the buffer size should be large enough and hence, an unacceptable delay will be introduced. The talk-spurt losses could be reduced to acceptable levels by allocating an appropriate service capacity *C* at the AP. Thus, the bufferless fluid-flow approach is applied for voice packets loss probability evaluation [7], and hence

(3) $$PPL = \frac{\sum_{i=\lceil n^* \rceil}^{N} (i - n^*) \cdot P_i}{N \cdot \alpha},$$

where α is the activity factor (the probability that a single ON-OFF source is active). Typical values of α can range from 0.35 to 0.45 [8]. The amount of network resources, also referred to as transmission resource units, is denoted by n^*. It represents the maximum number of active voice calls that can be simultaneously served without any packet losses. The probability P_i expresses that i sources are active (hold in ON state) out of N admitted ones and is given by the binomial distribution

(4) $$P_i = \binom{N}{i} \cdot \alpha^i \cdot (1-\alpha)^{N-i}.$$

Since the service capacity is influenced by the instantaneous SNR, n^* is not constant anymore. The spectral efficiency (SE) achieved at the physical layer depends on both the modulation scheme in operation and channel quality. It is tightly coupled with the service capacity, as well. For an *M*-ary QAM the spectral efficiency can be expressed by the well-known Shannon's capacity equation degraded by the SNR gap Γ [9]:

(5) $$SE(\gamma) = \log_2(1 + \gamma / \Gamma),$$

where $\Gamma = -\ln(0{,}5 \cdot BER)/1{,}5$. The Bit Error Rate (BER) can be calculated either analytically or by means of simulation analysis [10]. It is obvious the best AP throughput is achieved at the highest SNR. Thus, the above-mentioned relation of SE can be turned into normalized spectral efficiency

(6) $$\rho(\gamma) = SE(\gamma) / \max\{SE(\gamma)\}.$$

The maximum SE for an *M*-ary QAM is $\log_2 M$ data bits over a single transmission symbol per unit of bandwidth. The forthcoming analysis requires the total SNR range to be partitioned into *L* non-overlapping consecutive intervals with boundary points $\gamma \in [\gamma_n, \gamma_{n+1})$. For each interval, the average normalized spectral efficiency can be derived as

(7) $$\bar{\rho}(\gamma_n) = \frac{1}{\gamma_{n+1} - \gamma_n} \cdot \int_{\gamma_n}^{\gamma_{n+1}} \rho(\gamma) d\gamma.$$

In order to be computationally tractable, an approximated expression of (6) can be derived by polynomial fitting. The fitting parameters are modulation dependent and are obtained with respect to higher accuracy achievement.

Hence, the transmission resource units n^* are obtained as

(8) $$\lceil n^* \rceil = \bar{\rho}(\gamma_n) \cdot n \,,$$

where n denotes the transmission resource units available at the highest spectral efficiency for a particular modulation scheme in operation.

Similar to the above considerations, the wireless channel's dynamic behavior results in varying packet error rate. If each bit inside the voice packet has the same BER and bit errors are uncorrelated, the PER can be determined by BER through

(9) $$PER = 1 - (1 - BER)^{N_p}$$

for a packet containing N_p bits. The PER expression (9) is accurate for uncoded *M*-ary QAM even with large constellations [11]. However, this is not the case for coded *M*-ary QAM where the exact BER and PER can be obtained by means of simulations, since the PER in (9) is no longer accurate for such kind of transmissions (exact closed-form expression is not available).

Taking into account both packet loss probability and transmission error probability, for an average received SNR $\bar{\gamma}$, as well as wireless channel behavior, the average probability of unsuccessful reception of voice packets is obtained as

(10) $$P_{tot}^{\bar{\gamma}} = \sum_{\gamma=0}^{L} P_{loss}^{(\gamma)} \cdot \Pr(\gamma) = \sum_{\gamma=0}^{L} \left\{ 1 - \left(1 - \frac{\sum_{i=\lceil n^* \rceil}^{N} (i - n^*) \cdot P_i}{N \cdot \alpha} \right) \cdot \left(1 - PER(\gamma,\, N_p)\right) \right\} \cdot \Pr(\gamma) \,.$$

The last term in (10) $\Pr(\gamma)$ represents the probability that the instantaneous SNR falls within the range $\gamma \in [\gamma_n, \gamma_{n+1})$ and is expressed as

(11) $$\Pr(\gamma) = \int_{\gamma_n}^{\gamma_{n+1}} p_{ch}(\gamma) d\gamma = \frac{\Gamma\left(m, \frac{m \cdot \gamma_n}{\bar{\gamma}} \right) - \Gamma\left(m, \frac{m \cdot \gamma_{n+1}}{\bar{\gamma}} \right)}{\Gamma(m)} \,,$$

where $\Gamma(m, x)$ is the complementary incomplete Gamma function.

NUMERICAL RESULTS

This section deals with numerical results of the voice traffic performance analysis over non-ideal wireless environments. Time-varying nature of the wireless channel is modeled by (1) and is expressed by the average received SNR $\bar{\gamma}$. It is assumed that $\bar{\gamma}$ ranges between 5 and 30 dB. Performance analysis has been done for a particular modulation scheme – either QPSK or 16-QAM. The Bit Error Rate (BER) analysis for either modulation scheme has been carried out with the help of a BERTool implemented into the MATLAB environment [12]. The analysis could be done via theoretical, semianalytic, or simulation-based approach. The Ricean channel model is considered in this study where the Ricean *K* factor is mapped to the Nakagami fading parameter *m*.

Fig. 1 depicts the overall average probability of unsuccessful voice packets reception at the AP as a function of the average received SNR and the modulation method. The voice traffic is generated by 100 homogeneous and independent voice sources with VAD. Each packet incorporates both signaling (header) and payload fields. The latter depends on the voice coding scheme. If G.729 (8 kbps) codec is used, the overall packet size would be 78 Byte (624 bit), which includes RTP, UDP, IP, and MAC headers. This is the case when the payload field carries single voice frame of 20 Bytes. If four consecutive frames are wrapped into a payload, the packet size becomes approximately 1100 bit. It could be seen that the packet size does not significantly affect the probability of unsuccessful reception. The Ricean *K* factor is a part of the statistical description of the Ricean distribution and represents the ratio between the power in the line-of-sight component and the power in the multipath components. The more non-line-of-sight components occur (lower fading parameter *m*), the more unreliable is getting the voice service provisioning. This is also the case when a higher order modulation is used under poor channel conditions, which would lead to unacceptable levels of voice packet losses.

Beside the wireless channel dynamic, another source of packet losses comes from improper allocation of transmission resource units at the AP. This is clearly demonstrated on Fig. 2. By adjusting the amount of transmission resource units, allocated to the aggregated VoIP traffic flow, the overall packet losses could be reduced significantly to an acceptable level. It is important to note that further increase of this parameter cannot lead to considerable improvements due to wireless channel influence, i.e. the PER bound has been reached.

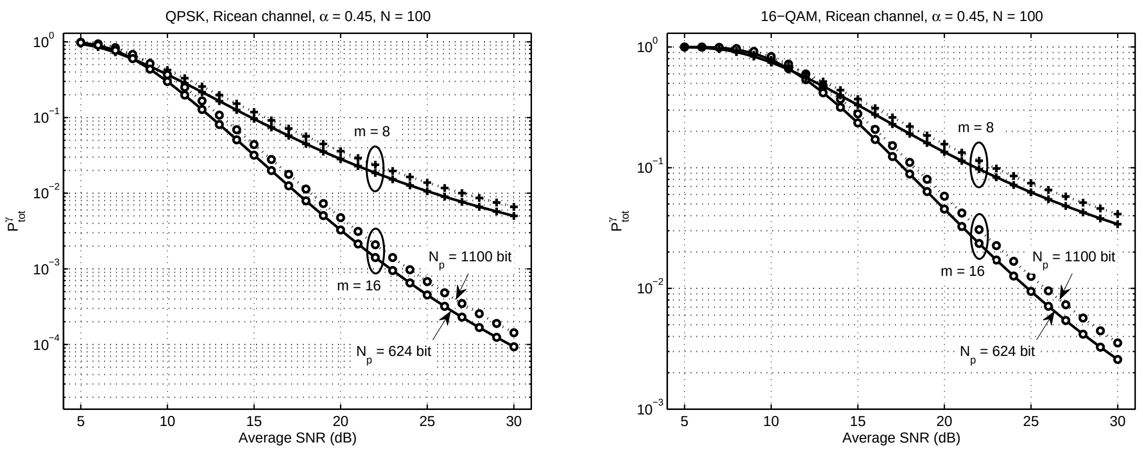

Fig.1 Probability of unsuccessful voice packets reception as a function of the average received SNR

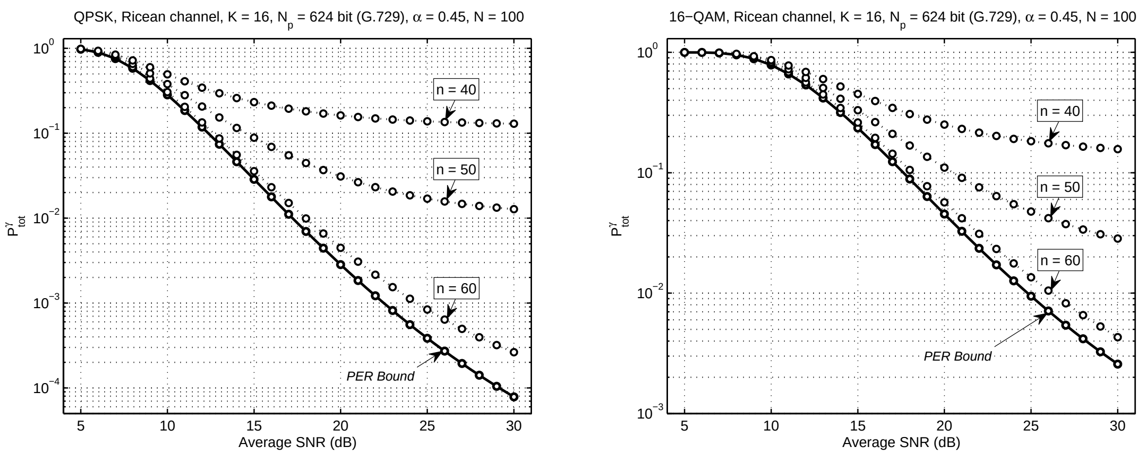

Fig.2 Probability of unsuccessful voice packets reception as a function of the average received SNR and the number of transmission resource units

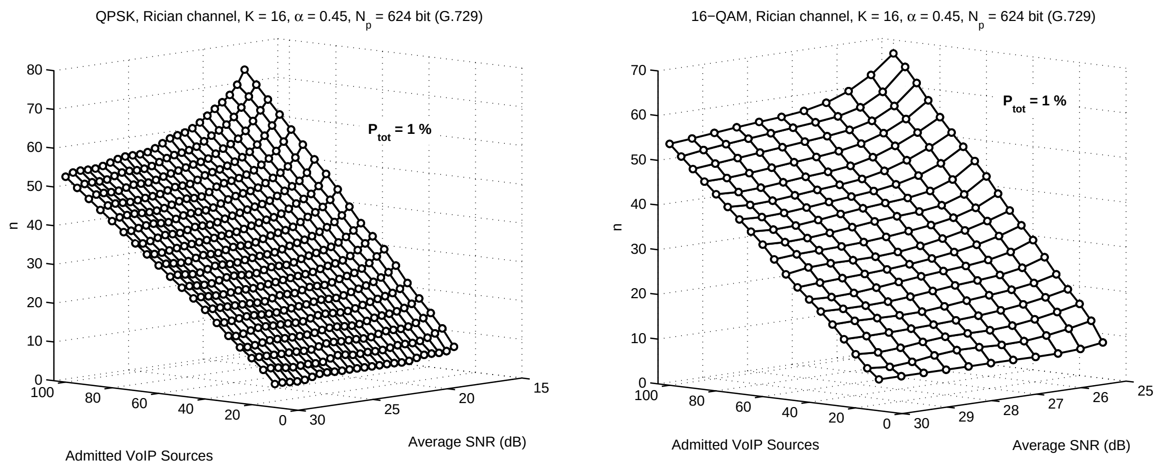

Fig.3 Network resource units dimensioning fulfilling a packet losses threshold

The call admission control is a crucial function in the contemporary IP-based networks serving multimedia traffic flows. It provides a flexible tool for resource management and QoS guarantee. In this article, the call admission decision is based on a simple rule: a new voice connection (call) is accepted if the amount of available transmission resource units is allocated such that the overall probability of unsuccessful voice packets reception fulfills a predefined threshold value. Fig. 3 shows an AP's resource units dimensioning with typical values of the overall packet losses threshold and activity factor α. Since QPSK is more robust than 16-QAM, it allows voice connections under poor channel conditions to be established satisfying the packet losses threshold. Of course, this is possible at the cost of more network resources to be allocated. On the other hand, 16-QAM does not efficiently utilize allocated network resources due to its low noise immunity.

CONCLUSION

This article investigates voice traffic transmission over wireless links. It focuses on a quantitative analysis of a CAC mechanism with respect to the voice traffic characteristics, packet size and the modulation scheme employed. The analytical method can be applied for resource estimation and dimensioning of wireless access networks taking into consideration time-varying behavior of wireless transmission medium.

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ОБСЛУЖВАНЕ НА ТРАФИК ОТ РЕЧ В БЕЗЖИЧНИ МРЕЖИ

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Ключови думи: *Безжичен канал за връзка, Безжични мрежи за достъп, Глас през IP мрежи, Детектор на речева активност, Качество на обслужване, Количествен анализ, Управление на достъпа на повикванията*

Резюме: *Оразмеряването на ресурсите в безжичните мрежи за достъп е първостепенна задача, която има за цел да се разработят модели за точна количествена оценка на мрежовия ресурс. За да се гарантира качество на обслужване (QoS) при предаване на трафик от реч в IP мрежи, се налага разработване на механизми за управление на достъпа на повикванията. Настоящата статия има за цел да представи подход за оразмеряване на механизъм за управление на достъпа и разпределяне на ресурси в безжични мрежи. Решението за достъп се формира на базата на общата вероятност за неуспешно приемане на речевите пакети, в резултат на ограничения мрежови ресурс в точката за безжичен достъп и динамиката на безжичния канал за връзка.*