Listen-Before-Talk Versus Treating Interference as Noise for Spectrum Sharing

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***Abstract*— The spectrum sharing problem between heterogeneous networks that are not interoperable is considered. Two strategies for interference management are studied. First, by treating the possible interference from adjacent transmitters as noise, each transmitter can achieve a certain information rate which merely depends upon the channel quality, but does not depend upon the burstiness of packets arrival. Second, in an ideal listen-before-talk (LBT) strategy, where perfect sensing between transmitters is assumed, the stable achievable rates of the pairs of transceivers are analyzed, and shown to exhibit interaction and dependence upon the bursty arrivals of packets. It is revealed that, for receivers in certain regions which are exposed to strong interference and low traffic burstiness, LBT performs better; while in the other regions with decreased interference level or higher traffic loads, treating interference as noise leads to better performance.**

I. Introduction

The proliferation of wireless devices and applications has generated a huge demand for bandwidth that is expected to grow well into the future. Unfortunately bandwidth is very scarce and hence an efficient utilization of this resource is crucial. In order to efficiently utilize the limited spectrum, different systems will have to share the same channel.

Dynamic spectrum access or spectrum sharing refers to techniques in which different systems try to opportunistically access the channel while managing the interference among them. Interference is the major limitation for spectrum sharing because if it is not properly managed it can limit the capacity that the systems can achieve. The problem becomes more severe when networks sharing the medium are heterogeneous. The main challenge to communication in such heterogeneous environments lies in striking a balance between the conflicting goals of minimizing the interaction among the users and maximizing the performance that each system can achieve.

Heterogeneous networks can have different air-interfaces, different PHY, MAC and completely different operating parameters. For such heterogeneous networks, interference management techniques that depend on explicit message exchange might not be feasible, especially when the networks are designed according to different standards. Consequently, sophisticated information-theoretic techniques such as rate-splitting and interference alignment are excluded, since those require global knowledge of the channel and the codebooks used by the different networks [1].

The basic two-by-two inference channel model is shown in Fig. 1, and models with more user pairs follow analogously. Traffic burstiness is taken into account in studying the interference channel. There are relatively few works considering jointly using information theory and queuing theory, and the relation between them is still far from fully revealed [2], [3]. Two spectrum sharing schemes are analyzed. In the first scheme, the interference at each receiver is treated as noise. Since the traffic is assumed bursty, a worst case scenario is considered in which the rise above the thermal noise floor caused by interference is always calculated and taken into account in the link adaptation. For such scenario, the achieved rates becomes independent from the traffic burstinees. The second scheme considered is when both transmitters use a listen-before-talk contention mechanism, so that each transmitter only sends when the channel is sensed to be idle or it wins the contention when both transmitters have packets in queues. For such scenario, there is an interaction between the stable achievable rates by each user since the achievable rates depend on the source burstiness.

The rest of the paper is organized as follows. In Section II, the system model is described. The performance analysis of listen-before-talk and treating interference as noise is considered in Section III. Numerical results are presented in Section IV, and finally conclusions are presented in Section V.

II. Model

An interference channel model is considered as shown in Fig. 1 in which terminal T1 transmits to terminal R1,

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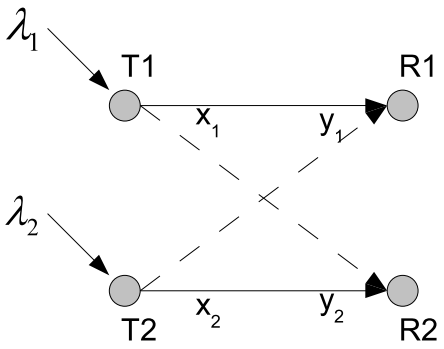


Fig. 1. System Model.

while terminal T2 transmits to terminal R2 over the same wireless channel. The channel is slotted, and all transmitting terminals are assumed to have infinite buffers. The arrival process at each transmitting terminal's queue is independent and identically distributed (i.i.d.) from one slot to another, and the arrival processes are mutually independent between the two transmitting terminals. Therefore the arrival processes are Bernoulli with the average arrival rates given by λ_1 and λ_2 at terminals T1 and T2, respectively.

The received signals at terminals R1 and R2 can be modeled as

$$\begin{aligned} y_1 &= \sqrt{P_1 L_{11}} h_{11} x_1 + n_1 + I_2 \sqrt{P_2 L_{21}} h_{21} x_2 \\ y_2 &= \sqrt{P_2 L_{22}} h_{22} x_2 + n_2 + I_1 \sqrt{P_1 L_{12}} h_{12} x_1 \end{aligned} \tag{1}$$

where P_i is the transmit power from terminal Ti and x_i is the transmitted signal (normalized to unit power). The term L_{ij} denotes the path-loss attenuation from terminal Ti to terminal Rj, modeled using a Hata path-loss model as described next. The channel h_{ij} between terminals Ti, Rj is modeled as a Rayleigh flat fading channel with unit variance. The terms n_i denotes zero-mean additive white complex Gaussian noise with variance N_o. The indicator function I_i models whether there is interference or not, and depends on the sharing protocol used.

The Hata model for path-loss in urban areas is given by [4]

$$\begin{aligned} L(d) = 69.55 + 26.16 \log f_c - 13.82 \log l_{tx} - a(l_{rx}) + \\ (44.9 - 6.55 \log l_{tx}) \log d, \end{aligned} \tag{2}$$

where $L(d)$ is the path loss in dB, f_c is the carrier frequency in MHz, l_{tx} and l_{rx} are the heights of transmit and receive antennas in meters, respectively, and d is the link distance in km. In (2), $a(l_{rx})$ denotes a mobile antenna correction factor, which for large cities is given by

$$a(l_{rx}) = 3.2 \left(\log \left(11.75 l_{rx}\right)\right)^2 - 4.97 \ f_c \geq 300 MHz. \tag{3}$$

A modified Shannon capacity formula is used for link adaptation in which a gap to capacity is considered to model the non-ideal coding and modulation schemes used, and an upper limit on the achievable spectral efficiency is used to model the highest packet format used by the system. The link adaptation formula is given as follows

$$C(\rho) = \min \left(\log \left(1 + \gamma \rho\right), C_{max}\right), \tag{4}$$

in which γ models the gap to capacity and C_{max} is the spectral efficacy of the highest packet format. In (4), ρ denotes signal to noise and interference ratio. Terminals are assumed to be transmitting at rates given by (4), and given a fixed frame size of T seconds and bandwidth B Hz, the size of the transmitted packets in terms of the number of bits per packet differs according to the achievable link capacity. Since from a queuing theory prospective we want to treat packets as entities entering and leaving the queue, the packet size S is given in terms of the service rate in bits per seconds (bps) and the frame duration as follows $B \cdot C(x) \cdot T$. Thus the packet arrival rate per frame can be written in terms of the data arrival rate in bps as follows

$$\lambda = \frac{\Lambda}{BC(x)}, \tag{5}$$

where Λ is the data arrival rate in bps.

III. Stable Throughput Analysis

A. Treating Interference as Noise

Perhaps the simplest way of accommodating multiple transmitters is letting each treat the others's potential transmission as non-intelligent interference, *i.e.*, noise. In doing so, a transmitter attempts to transmit at a fixed rate whenever its queue is non-empty, regardless of the status of the other transmitter. Therefore, the whole queueing system remains stable as long as the data arrival rate is strictly lower than the link service rate, for each transceiver pair.

The service rates of the two terminals can be made interdependent if sophisticated scheduling is allowed between the transceiver pairs. Due to practical considerations, in this paper we take a conservative approach, which demands only the minimum amount of channel knowledge. Specifically, we assume that each receiver knows the channel fading realization of the link from its corresponding transmitter, and only knows the fading variance of the link from the interfering transmitter. Furthermore,

since there is no effective way for a transmitter to know whether the other is transmitting, to ensure that its own transmission can be reliably decoded at a fixed rate, we take a conservative approach, *i.e.*, transmitting at a rate assuming that the interference always exists.

From the above consideration, the service rates in bps can be expressed as

$$\mu_1 = B \cdot C\left(\mathrm{SINR}_1\right) \tag{6}$$

$$\mu_2 = B \cdot C\left(\mathrm{SINR}_2\right), \tag{7}$$

where $C\left(\cdot\right)$ is defined in (4), and the worst-case signal-to-interference-plus-noise ratio (SINR) $\mathrm{SINR}_{1,2}$ are given by

$$\mathrm{SINR}_1 = \frac{P_1 L_{11} |h_{11}|^2}{P_2 L_{21} \mathrm{var}[|h_{21}|^2] + N_o} \tag{8}$$

$$\mathrm{SINR}_2 = \frac{P_2 L_{22} |h_{22}|^2}{P_1 L_{12} \mathrm{var}[|h_{12}|^2] + N_o}. \tag{9}$$

We note that in deriving $\mu_{1,2}$, we have taken the following conservative measures in order to ensure reliable communication regardless of channel and queue states. First, each transmitter assumes that the other transmitter is always transmitting so creating interference. Second, in dealing with the interference, since the fading coefficient from the interfering transmitter is not known by the receiver, we treat the faded thus non-Gaussian interference random variable as Gaussian, to yield a lower bound to achievable rates. Third, each transmitter is not allowed to adaptively perform short-term power/rate control, due to the lack of channel knowledge at the transmit side.

For a general M-user scenario, the service rates can analogously be expressed as

$$\mu_j = B \cdot C\left(\mathrm{SINR}_j\right), \quad j = 1, \ldots, M, \tag{10}$$

where

$$\mathrm{SINR}_j = \frac{P_j L_{jj} |h_{jj}|^2}{\sum_{i=1, i \neq j}^{M} P_i L_{ij} \mathrm{var}[|h_{ij}|^2] + N_o}. \tag{11}$$

B. Listen-Before-Talk

Consider a time slotted system in which a packet is transmitted in each time frame. At the beginning of each frame, N time slots are used for contention between systems. The contention time is assumed to be smaller than the frame length and hence its impact on the throughput is negligible. Each transmitter generates a random variable with uniform distribution between 1 and N and starts a counter at the beginning of each frame. The transmitter listens to the medium and decrements its counter each time slot if the medium was idle otherwise the transmitter abandons transmission in this time slot. If the counter expires and no other transmission is heard, the transmitter will transmit the packet at the head of its queue. First we consider the case of two pairs of nodes.

1) Two User Scenario: The evolution of the queue for transmitter T_i, $i = 1, 2$, can be modeled as

$$Q_i^{t+1} = \left[Q_i^t - Y_i^t\right]^+ + X_i^t \tag{12}$$

where Q_i^t is the Queue length of the i-th transmitter at time t, X_i^t denotes the packet arrival process to the queue and is modeled as a Bernoulli random variable with probability λ_i, and Y_i^t denotes the service process, or equivalently the possible (virtual) departure process. By virtual we mean that Y_i^t could equal 1 even if the queue does not have any packets to transmit [6]. The function $[\cdot]^+$ is defined as follows

$$[x]^+ = \begin{cases} x, & \text{if } x > 0 \\ 0, & \text{if } x \leq 0 \end{cases} \tag{13}$$

The service process for the first queue Y_1^t can be modeled as follows

$$Y_1^t = \mathbf{1}\left[Q_2^t = 0\right] + \mathbf{1}\left[\left\{Q_2^t > 0\right\} \bigcap \left\{W_1^t\right\}\right] \tag{14}$$

where, $\mathbf{1}[\cdot]$ is the indicator function, W_i^t denotes the event that transmitter T_i wins the contention at frame t. Assuming that both transmitters can hear each other, i.e. no hidden node, the probability of the event W_i^t can be shown to be given by

$$P\left(W_i^t\right) = \frac{1}{2}\left(1 - \frac{1}{N}\right). \tag{15}$$

If we ignore the capture effect in the following analysis, i.e. any collision leads to losing the transmitted packets, the achievable rates calculated will represent lower bounds on what can be achieved. On the other hand, if the collision probability $1/N$ is ignored, the calculated rates will be upper bounds for the achievable throughput. These bounds are tight if the number of contention time slots N is large enough.

Let us calculate the expected value of the service process Y_i^t. Given that the queue of the second transmitter is stable, from (14) $\mathrm{E}\left[Y_1^t\right]$ can be calculated as

$$\mathrm{E}\left[Y_1^t\right] = \left(1 - \frac{\lambda_2}{\mathrm{E}\left[Y_2^t\right]}\right) + \frac{1}{2}\left(1 - \frac{1}{N}\right)\frac{\lambda_2}{\mathrm{E}\left[Y_2^t\right]}, \tag{16}$$

where $\frac{\lambda_2}{\mathrm{E}[Y_1^t]}$ is the probability that the second queue is not empty [5]. Rearranging the terms in the above equation, the average service rate for T_1 can be calculated as

$$\mathrm{E}\left[Y_1^t\right] = \left(1 - \frac{1}{2}\left(1 + \frac{1}{N}\right)\frac{\lambda_2}{\mathrm{E}\left[Y_2^t\right]}\right). \tag{17}$$

Similarly, the average service rate for T_2 can be calculated as

$$\mathrm{E}\left[Y_2^t\right] = \left(1 - \frac{1}{2}\left(1 + \frac{1}{N}\right)\frac{\lambda_1}{\mathrm{E}\left[Y_1^t\right]}\right). \tag{18}$$

Writing the above equations in terms of achievable throughput Λ_1 and Λ_2 in bits per second, we get

$$\mu_1 = B \cdot C_1 \left(1 - \frac{1}{2}\left(1 + \frac{1}{N}\right)\frac{\Lambda_2}{\mu_2}\right) \tag{19}$$

and for terminal 2

$$\mu_2 = B \cdot C_2 \left(1 - \frac{1}{2}\left(1 + \frac{1}{N}\right)\frac{\Lambda_1}{\mu_1}\right), \tag{20}$$

where C_1 C_2 are the channel capacities for the pairs 1 and 2, respectively and are calculated as $C_i = B\log\left(1 + \mathrm{SNR}_i\right)$, and SNR_i is the SNR of the signal received at receiver i and is given by

$$\mathrm{SNR}_i = \frac{P_i L_{ii} |h_{ii}|^2}{N_o}. \tag{21}$$

In (19) and (20), μ_i is the achievable service rate in bits per sec and is given by

$$\mu_i = B \cdot C_i \mathrm{E}\left[Y_i^t\right]. \tag{22}$$

The interdependency between the queues in the two systems is clear from (19) and (20). Solving the two equations, the service rate for the first terminal μ_1 can be shown to be the solution of the following equation

$$\begin{aligned} C_2\mu_1^2 - \left(C_2\frac{1}{2}\left(1 + \frac{1}{N}\right)\Lambda_1 - C_1\frac{1}{2}\left(1 + \frac{1}{N}\right)\Lambda_2\right. \\ \left.(+C_1C_2)\,\mu_1 + C_1C_2\frac{1}{2}\left(1 + \frac{1}{N}\right)\Lambda_1 = 0, \end{aligned} \tag{23}$$

Equation (23) is a quadratic equation with two roots that are given by

$$\begin{aligned} \mu_1 = B \cdot \frac{\left(C_2\frac{1}{2}\left(1 + \frac{1}{N}\right)\Lambda_1 - C_1\frac{1}{2}\left(1 + \frac{1}{N}\right)\Lambda_2 + C_1C_2\right)}{2C_2} \\ \pm B \cdot \frac{\sqrt{\Phi}}{2C_2}, \end{aligned} \tag{24}$$

where

$$\begin{aligned} \Phi = \left(C_2\frac{1}{2}\left(1 + \frac{1}{N}\right)\Lambda_1 - C_1\frac{1}{2}\left(1 + \frac{1}{N}\right)\Lambda_2 + C_1C_2\right)^2 \\ - 4C_1C_2^2\frac{1}{2}\left(1 + \frac{1}{N}\right)\Lambda_1. \end{aligned} \tag{25}$$

Substituting $\Lambda_2 = 0$ in the above equation, the negative root leads to the solution $\mu_1 = \Lambda_1$ and the positive root gives the solution $\mu_1 = B \cdot C_1$. Since when $\Lambda_2 = 0$ terminal 1 occupies the channel all of the time, its service rate will be determined by the channel capacity. Hence the positive root is the right solution and the other root should be discarded. The service rate for terminal 2 can be found similar to the above steps.

If we are interested in an equal grade of service (EGoS) type of applications, then all terminals should have the same served throughput. Hence we are interested in solving for the maximum served throughput at which the network is stable. Since this is a *maxmin* criteria, the solution will be at the point when the link with the minimum capacity is about to be unstable. Let us assume without loss of generality that $C_2 < C_1$. The at the solution for EGoS we must have

$$\mu_2 = \Lambda. \tag{26}$$

Substituting in (19) we get

$$\mu_1 = \frac{B}{2}\left(1 - \frac{1}{N}\right)C_1 \tag{27}$$

Substituting the value of μ_1 into (20) we get

$$\Lambda = B \cdot C_2 \left(1 - \left(1 + \frac{1}{N}\right)\frac{\Lambda}{\left(1 - \frac{1}{N}\right)C_1}\right). \tag{28}$$

Solving for Λ we get

$$\Lambda_{EGoS} = \frac{B}{\frac{1}{\min(C_1,C_2)} + \frac{N+1}{N-1}\frac{1}{\max(C_1,C_2)}}. \tag{29}$$

If the collision probability is ignored, which reduces to time sharing between the two systems, then the EGoS rate is given by

$$\Lambda_{EGoS} = \frac{B}{\frac{1}{C_2} + \frac{1}{C_1}}. \tag{30}$$

which is the harmonic mean of the capacities of the two channels.

2) Multiple User Scenario: For a multiple user scenario, the problem is more complicated since terminals queues are correlated. Consider M transmitters, the departure process for terminal 1 can be written as

$$Y_1^t = \mathbf{1}\left[\bigcap_{j=2}^{M}\left\{Q_2^t = 0\right\}\right] + \mathbf{1}\left[\bigcup_{j=2}^{M}\left\{Q_j^t > 0\right\}\bigcap\left\{W_1^t\right\}\right] \tag{31}$$

Since all queues are correlated, it becomes more complicated to write expressions for the average service rate because it is difficult to characterize the joint distribution of the queue sizes.

For simplicity, let us assume that $N >> M$ and hence the collision probability becomes very small. For such scenario it is easy to show that the maximum EGoS throughput is given by the harmonic mean of all the link capacities.

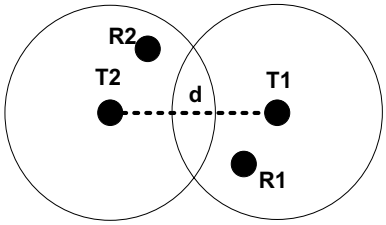
$$\Lambda_{EGoS} = \frac{B}{\sum_{j=1}^{M} \frac{1}{C_j}}. \quad (32)$$

IV. Numerical Results

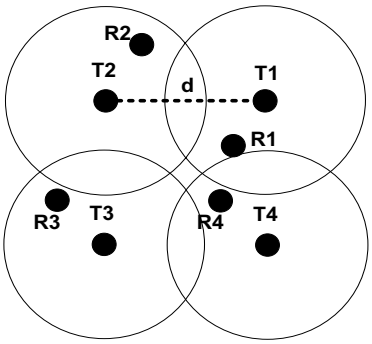
In this section, we numerically evaluate the performance of treating interference as noise and listen-before talk under different interference scenarios. First we consider the scenario with two pairs of terminals as in Fig. 2(a). The coverage area for each terminal is defined in terms of the minimum throughput that a cell edge user can maintain which is 0.5bps/Hz. 36dBm transmit EIRP is assumed at the transmitter. A 3dB gap to capacity is considered to model non-ideal modulation and coding at the transmitter, and the highest packet format used is 64QAM with code rate 1 which accounts to 6bps/Hz. The carrier frequency is taken to be 600MHz and the channel bandwidth is 5MHz. The antenna heights for terminals T1 and T2 is 12m and for terminals R1 and R2 is 2m. A 15 dB total fixed losses that model building penetration, cable and body losses is considered. Log normal shadow fading is modeled with standard deviation of 8dB. For these parameters and using the Hata model (2) the cell radius is found to be 0.23km.

R1 and R2 are dropped with uniform distribution inside the coverage area of terminals T1 and T2, respectively. The maximum stable throughput achieved is calculated for the EGoS scenario using (30) for listen-before-talk (LBT). For treating interference as noise (TIAN), the minimum of the capacities of the two links is calculated. Numerical results for the two terminal case are depicted in Fig. 3 in which the cumulative distribution function (CDF) of the maximum stable throughput is depicted. Different separation between the two transmit terminals are considered. The distance between the two terminals d is given as a function of the cell radius R. From the results, it is clear that there is a tradeoff between LBT and TIAN. For example, for $d = R$ 18% of the users are better off using TIAN than LBT. When the separation is increased to $2R$, this percentage increases to 47%.

Fig. 4 depicts the four terminals case when the four terminals are on a square as in Fig. 2(b). As clear from the figure, the tradeoff between the two schemes shifts in favor of LBT because of the higher interference levels seen with more terminals.



(a) Two Transmitters



(b) Four Transmitters

Fig. 2. Simulation Models.

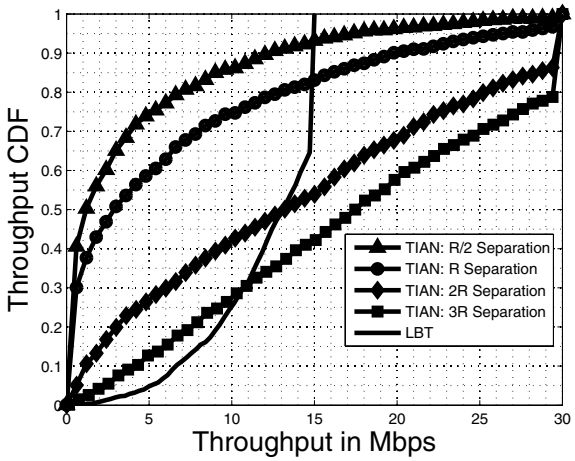


Fig. 3. CDF of the Throughput in Mbps for the two terminal case.

V. Conclusions

The performance of listen-before-talk and treating interference as noise has been analyzed under an information theoretic and queuing framework. Bursty traffic sources are considered and the achievable rates for both schemes are calculated. It is shown through analytical and numerical results that, within the same cell, some users have better performance when treating interference as noise while others have better performance when using contention based listen before talk. This not only de-

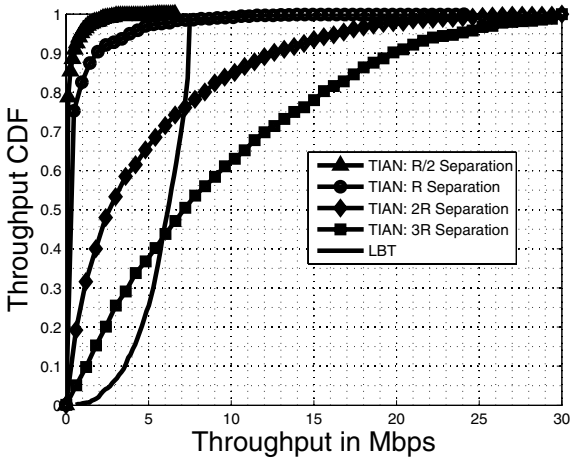


Fig. 4. CDF of the Throughput in Mbps for the Four terminal case.

pends on the interference level seen but also on the traffic burstiness. The tradeoff between the two schemes becomes more in favor of listen before talk when the number of terminals sharing the medium increases.

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