An Effective Resource Management Scheme for UWB Networks with Simultaneous Transmissions

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***Abstract*— This paper aims at an effective resource management scheme for ultra-wideband (UWB) networks where the inherent spread spectrum supports simultaneous transmissions. In specific, we present a transmission frame structure tailoring to the UWB characteristics, and develop a novel control message exchange procedure. Furthermore, we propose effective admission control and resource allocation algorithms to achieve high efficiency. The resource management scheme can solve the *near-sender-blocking problem* and alleviate the negative effect of long acquisition time in UWB transmissions. Extensive simulations demonstrate the superior performance of our proposed scheme.**

***Index Terms*— Admission control, ultra-wideband (UWB), quality of service (QoS), resource allocation.**

I. INTRODUCTION

TO support high-speed short-range wireless connections, ultra-wideband (UWB) has emerged as a promising technology in future wireless multimedia communications. A UWB system has a 10 dB bandwidth larger than 500 MHz, or has a 10 dB fractional bandwidth larger than 20%. Its commercial deployment has attracted significant attention recently, especially after the U.S. Federal Communications Commission (FCC) approved the use of the frequency band from 3.1 to 10.6 GHz by UWB indoor applications. It has been demonstrated that UWB technology has many promising advantages such as high bit rate, low power/interference, robustness to multipath fading, and localization capability. Typical UWB applications span from traditional multimedia applications to home/office networking and control, industrial maintenance, medical imaging, sensor networks, as well as Department of Defense (DoD) systems [1], [2].

In a UWB wireless network, the medium is shared among mobile nodes. To achieve desired quality of service (QoS) such as transmission accuracy, delay/jitter, throughput, and fairness, the radio resources should be managed in an effective and orderly manner [2]. The inherent spread spectrum in UWB can support simultaneous transmissions, with an appropriate pseudorandom sequence design and effective call admission control (CAC). In such a *multi-channel* case, two nearby transmissions do not collide, but generate interference to each other. In the literature, one major stream of UWB access control is IEEE 802.15.3 based, which is designed for short-range ad hoc connectivity in wireless personal area networks (WPANs). However, it is not explicitly designed for UWB-based multi-channel transmissions.

Power control has been considered as an effective way to combat multi-user interference and guarantee the required transmission accuracy at the receiver in traditional code-division multiple access (CDMA) cellular networks. However, it is challenging to frequently reconfigure power levels in UWB networks as the exchanges of control messages can be costly. To keep a low level of control message exchanges, a maximum sustainable interference (MSI)-based scheme is introduced in [3], which is basically a circuit-switching channel reservation. The MSI-based scheme may have a severe *near-sender-blocking problem* where a sender near the receiving node of another link may prohibit the admission of any new calls to the network (see Section III). An effective solution to address the near-sender-blocking problem is to use temporal exclusion mechanisms. Generally, temporal exclusion mechanisms can be categorized into two groups according to whether or not the time is slotted [4]: unslotted reservation and slotted reservation. Unslotted reservation is mainly designed for single-channel contention-based networks, and is not effective for UWB networks with multi-channel transmissions. In slotted reservation, time is partitioned into frames which are further divided into fixed-length slots. The mobile nodes contend to transmit in these slots. The distributed packet reservation multiple access (PRMA) [4] and five-phase reservation protocol (FPRP) [5] make slot reservation in a distributed manner for single-channel networks. Thus they are not suitable for UWB networks. Another big challenge in UWB communications is the long acquisition time required by the high-precision synchronization, usually varying from tens of microseconds to tens of milliseconds compared to microseconds in narrowband systems. The long acquisition time has been shown to degrade the achieved throughput severely [6], which should be considered in the resource management.

To address the above issues, this paper aims at developing an effective and efficient resource management scheme in UWB wireless networks with multi-channel transmissions. In specific, we present a transmission frame structure tailoring to

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TABLE I
SUMMARY OF IMPORTANT SYMBOLS

Symbol	Definition
C_i^k	cost function of link i at slot k
h_{ij}	channel gain from link i's transmitter to link j's receiver
M	number of slots in each time frame in our proposed scheme
MSI_i	MSI value of active link i
MSI_i^k	MSI value of active link i at slot k
$N(k)$	number of existing links at slot $k \in \{1, ..., M\}$
$P^{\max}$	maximum transmission power of a node
P_i (P_i^k)	transmission power of link i (at slot $k \in \Omega_i$)
p_i^k	transmission power of link i at slot k under the hypothesis that the new call request (for link i) selects slot k for service
Q_i^k	penalty function of resource consumption by link i at slot k
R_i^e	effective rate of link i, i.e., the total achieved rate seen by the user of link i
R_i (R_i^k)	transmission rate of link i (at slot $k \in \Omega_i$)
R_i^t	target rate of link i
r_i^k	transmission rate of link i at slot k under the hypothesis that the new call request (for link i) selects slot k for service
T_f	the pulse repetition time in TH-UWB
T_{frame}	length of a frame in our proposed scheme
T_{slot}	length of a time slot in our proposed scheme
U_i^k	utility function of link i at slot k
η	the background noise energy plus non-UWB interference energy
σ^2	a parameter depending on the shape of the pulse in TH-UWB
γ_i	required SINR value for link i
Ω_i	the set of slots ($\subseteq \{1, 2, ..., M\}$) in each frame from which link i has gained services
ξ	the acquisition overhead ratio (i.e., the fraction of time for acquisition in an slot if acquisition is needed)

the UWB characteristics, and develop a novel control message exchange procedure. Furthermore, we propose effective admission control and resource allocation algorithms to achieve high efficiency. The resource management scheme can solve the near-sender-blocking problem and alleviate the negative effect of long acquisition time in UWB transmissions.

The rest of the paper is organized as follows. The system model is given in Section II. The MSI-based scheme is discussed in Section III. Sections IV, V, and VI present the time frame structure, the admission control algorithm, and the resource allocation algorithm, respectively, followed by the performance evaluation in Section VII. Finally the conclusion remarks are given in Section VIII. As many symbols are used in this paper, Table I summarizes important ones.

II. System Model

UWB is mainly designed for short-range wireless communications [7]. We consider a UWB network covering a small area, which supports peer-to-peer single-hop communications with low mobility. Each node can hear any other node's transmission as long as it tunes to the transmission code and the received signal to interference-plus-noise ratio (SINR) exceeds a threshold. Simultaneous transmissions generate interference to each other. The maximum transmission power of each node is $P^{\max}$, which can be determined by the emission regulation and the energy consumption of the nodes. At the physical layer, the implementation of UWB transmissions can be achieved by pulse-based time-hopping (TH), pulse-based direct sequence (DS), or multiband orthogonal frequency division multiplexing (MB-OFDM). Our research is based on the TH-UWB with binary pulse position modulation (PPM), but the approach can be extended to DS-UWB.

A. Channel Model

The very short pulse transmissions in pulse-based UWB networks determine that UWB signal reception is robust to multipath fading [8]–[11]. Thus, similar to [12], we assume there is no fast fading, and the power at the receiver is attenuated due to path loss, i.e., the channel gain from link i's transmitter to link j's receiver can be represented as $h_{ij} = K \cdot d_{ij}^{-\theta}$, where K and θ are constants, and d_{ij} is the distance from link i's transmitter to link j's receiver.

In TH-UWB, the information bit is transmitted with a sequence of very narrow pulses (usually in the order of a nanosecond). Multiple access in TH-UWB can be achieved by assigning unique time hopping codes to different links. Each node is assigned a unique receiving code. The receiving code of the destination is used for any peer-to-peer transmission. Hence, each node only needs to monitor its own receiving code for the desired traffic [2]. It has been shown in [13] that the total interference from a large number of links can be approximated as Gaussian noise. Therefore, for a TH-UWB network with N active links, the SINR of link i can be represented as

$$\mathrm{SINR}_i = \frac{P_i h_{ii}}{R_i(\eta_i + T_f \sigma^2 \sum_{j=1, j\neq i}^{N} P_j h_{ji})}, \qquad i = 1, ..., N \tag{1}$$

where P_i denotes the average transmission power of link i's transmitter, R_i the bit rate of link i, η_i the background noise energy plus non-UWB interference energy, T_f the pulse repetition time, and σ^2 a parameter depending on the shape of the pulse [3]. Similar to [3], we use the Gaussian approximation as an example model for multi-user interference. However, other more accurate/practical models can be incorporated in our research when they are developed/validated.

B. QoS Requirements

QoS in radio resource management can be classified based on a hierarchy of two levels: bit-level and packet-level. Bit-level QoS is to ensure some degree of transmission accuracy, normally represented by an upper bound on bit error rate (BER) in transmissions. The BER guarantee can be achieved by satisfying a required SINR value γ_i for link i. The one-to-one mapping from BER to SINR depends on channel characteristics, modulation, channel coding, diversity, and receiver design. On the other hand, transmission rate (i.e., throughput), timeliness (i.e., delay and jitter), and fairness are the main consideration in packet-level QoS. In this paper, the QoS provisioning is to provide each link with a rate guarantee under the constraint of the required SINR bound, i.e., for each link i, the following inequality should hold

$$\frac{P_i h_{ii}}{R_i(\eta_i + T_f \sigma^2 \sum_{j=1, j\neq i}^{N} P_j h_{ji})} \geq \gamma_i. \tag{2}$$

An equivalent form of the inequality is

$$R_i \leq \frac{P_i h_{ii}}{\gamma_i(\eta_i + T_f \sigma^2 \sum_{j=1, j\neq i}^{N} P_j h_{ji})}. \tag{3}$$

The inequality gives the maximum achievable bit rate of link i with the constraint of SINR value γ_i. It is assumed that an adaptive rate can be achieved by changing the processing gain, e.g., via adapting the number of pulses for each symbol and/or the maximum time hopping shift, or via adaptive channel coding such as rate compatible punctured convolutional (RCPC) code. On the other hand, for each link, the maximum achieved bit rate should not exceed $1/T_f$ as there should be at least one pulse for each symbol. Therefore, for link i the achievable bit rate is

$$\min\{\frac{1}{T_f}, \frac{P_i h_{ii}}{\gamma_i(\eta_i + T_f\sigma^2 \sum_{j=1,j\neq i}^{N} P_j h_{ji})}\}.$$

In this research, we do not differentiate best-effort and real-time services. The QoS requirement for each link is a pre-specified service rate. However, service differentiation can be smoothly incorporated into our research. For instance, different traffic types can use different contention protocols when they contend for resource reservation.

III. MSI-Based Resource Management Scheme

It is important for a wireless network to keep a low level of control message exchanges. Hence, the frequent power reconfiguration in UWB networks should be avoided. In [3], an MSI-based scheme is introduced. Each link in the UWB network keeps an MSI, which is also referred to as the interference margin. The MSI denotes the additional tolerable interference while not violating the SINR requirement, i.e., for link i,

$$\frac{P_i h_{ii}}{R_i(\eta_i + T_f\sigma^2 \sum_{j=1,j\neq i}^{N} P_j h_{ji} + \mathrm{MSI}_i)} = \gamma_i \tag{4}$$

which leads to

$$\mathrm{MSI}_i = \frac{P_i h_{ii}}{\gamma_i R_i} - \eta_i - T_f\sigma^2 \sum_{j=1,j\neq i}^{N} P_j h_{ji}. \tag{5}$$

The MSI values of all the links are updated upon each new link admission, and should be nonnegative in order to keep the transmission accuracy of all the links. For multiple access in a multi-channel environment, each active link periodically announces its MSI value over a control channel. If a link's MSI is honored by all the other links, its transmission accuracy can be guaranteed. When a call request for a new link arrives at one node, according to MSI information of other links, the node determines whether or not it is feasible to have an assigned power level and a guaranteed rate. The MSI-based scheme uses a kind of circuit-switching channel reservation. Each link reserves a code channel, and a new link is required not to violate the QoS of existing links. However, it may have severe *near-sender-blocking problem*, as demonstrated in Fig. 1. At the beginning, there are two links, links 1 and 2, each with transmission power $P^{\max}$ at the sender. Subsequently link 3 becomes active. As the sender of link 3 is close to the receiver of link 1, it may generate significant interference to link 1's receiver. Thus the MSI of link 1 may be largely reduced (even to zero). It is very difficult for the network to admit another link because of no sufficient MSI for link 1, even though links 2 and 3 have large MSIs.

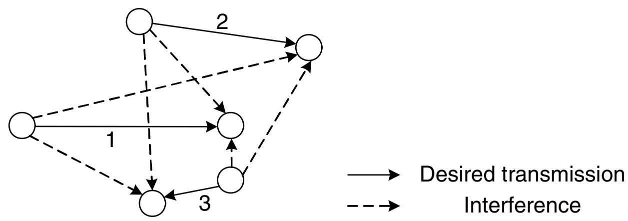

Fig. 1. The near-sender-blocking problem in MSI-based scheme.

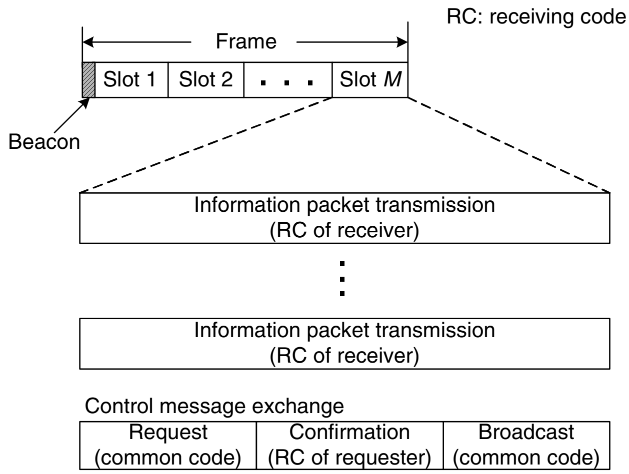

Fig. 2. The time frame structure for the proposed scheme.

Further, for the MSI-based scheme, some information exchanges are needed such as the MSI values, location, and transmission power of the active links. In [3], a control channel is used by all the senders. However, the exchange overhead is very large when the number of nodes is large, since each node needs to update and broadcast its information upon admission of each new link in the network. In addition, the broadcasts from different nodes may collide with each other. It is also difficult for a potential transmitter to collect complete information (i.e., the MSI, location, and power level) of all the existing links, thus very likely leading to service degradation.

An effective solution to address the near-sender-blocking problem is to use a temporal exclusion mechanism. As shown in Fig. 1, if links 1 and 3 transmit in different periods, the significant reduction of link 1's MSI can be avoided. Accordingly, we propose a resource management scheme for UWB wireless networks with multi-channel transmissions. The frame structure, admission algorithm, and resource allocation algorithm of the proposed scheme are detailed in the following three sections, respectively.

IV. Time Frame Structure and Control Message Exchanges

The frame structure for the proposed scheme is shown in Fig. 2. Each frame starts with a beacon, followed by M time slots. The beacon is to indicate the beginning of a frame. In each slot, multiple simultaneous transmissions are supported, for information packets and control message exchanges. For effective control message exchanges, among all the active senders at a slot, one is selected to act as the *slot head* (the selection procedure will be discussed later). The slot head is responsible to collect information at this slot and broadcast to potential new senders during its period of duty. As shown in Fig. 2, the control message exchanges are performed in parallel with other information packet transmissions. The exchange procedure includes three phases:

- Request phase: If a potential sender selects a slot to transmit, it will send a request at the request phase of the slot (by a pre-specified request common code). If the request is not transmitted successfully (i.e., no confirmation is received in the confirmation phase), the sender re-sends at the request phase of the slot in next frames with a probability p.
- Confirmation phase: Upon a successful reception of a new call request, the slot head responds with a confirmation via the receiving code of the requester.
- Broadcast phase: The slot head broadcasts (by a pre-specified broadcast common code) the MSI, location, and transmission power information of the existing links at the slot.

It can be seen that, because of UWB's capability in supporting multiple simultaneous transmissions, the time frame structure does not need an extra control slot for channel request as that in [5].

Consider a UWB network with no active nodes at the beginning. When a node has a call request, it first detects the beacon. If no beacon is detected, the node assumes that all the channels are idle, transmits a beacon, picks up the first one or more slots (depending on its rate requirement), and acts as the head for the slots. In each of the slots, the node not only transmits to its receiver (using the receiver's receiving code[1]), but also transmits the location, MSI, and transmission power information of all the active links at the slot via the broadcast common code. The beacon is always sent by the head of the first slot in a frame. The beacon can be heard by all the nodes. Synchronized to the received beacon, each node can generate a frame structure. When another node (a potential sender) has a call request for a new link, it first listens to the broadcast channel in all the slots, and collects the MSI, location, and transmission power information of each active link at each slot. It then selects (based on criterion discussed in Sections V and VI) a slot as its target slot, and sends a request at the request phase of the slot. There are two possible outcomes of the request. 1) If the request is received successfully by the slot head, the slot head responds with a confirmation via the requester's receiving code. Upon the reception of the confirmation, the requested new link will be admitted in the slot, and the new link sender becomes the new slot head in order not to pose all computation complexity on a single slot head. As the broadcast message from the old slot head contains all required information for a slot head, there is no extra overhead for information transfer between the old and new slot heads. The new slot head updates the MSI values of all the existing links, and broadcasts the MSI values, location and transmission power information of all the links (including its own link). An alternative solution is to fix the slot head for each slot, though the computation complexity and power consumption (in request reception, confirmation, and information broadcasting) of the fixed slot head may be large. Our proposed scheme can still be applied to this alternative solution with minor modifications. Generally, the novelty of the proposed *slot head rotation* mechanism lies in that: i) the computation burden and the power consumption of the slot head are fully distributed to the admitted senders, which is important to UWB devices with limited power supply, and ii) no handover command is needed. 2) If no confirmation is received (e.g., when at least one other node also sends a request simultaneously) for a new link request, the call sender re-sends the request at the same slot of the following frames with a probability p until a confirmation is received[2]. We use a heuristic method to design the parameter p. In the request phase, if the slot head detects a collision, it broadcasts a `PRE_COL` message in the broadcast phase; if it receives a request successfully, the slot head broadcasts a `PRE_SUC` message; if it detects no transmissions, the slot head broadcasts a `PRE_IDLE` message. When a node requests a slot, it first transmits its request with a probability 1 in the request phase. If failed, at the same slot of the subsequent frame the sender

- selects p uniformly from a set $\{p_1, p_2, .., p_m\}$ (where $0 < p_1 < p_2 < ... < p_m < 1$) if `PRE_COL` is received at the previous broadcast phase of the slot;
- sets $p = p_m$ if `PRE_SUC` or `PRE_IDLE` is received at the previous broadcast phase of the slot.

Then the node transmits a request with probability p at the request phase until a confirmation is received.

After a call is admitted into a slot, its rate requirement may not be satisfied. In this case, the call sender listens to the broadcast channel in other slots again, chooses a new target slot, and contends in the request phase until success. If a call is admitted to a slot, its sender can transmit at the same slot of subsequent frames until the call is completed.

If a call is completed, the sender of the link will contend at the request phase of its serving slots to send a `CALL_FINISH` message in the same way as sending a call request. When the sender contends successfully, the slot heads respond with a confirmation, and update the MSI information in their slots. If the sender of the link is the slot head of one or more serving slots, it does not need to contend in the request phase, but rather updates MSI values and takes the responsibility of slot heads until new slot heads appear.

Through the slot heads, the drawback of information exchange in the MSI-based scheme (as discussed in Section III) can be avoided effectively.

In order to make the estimation of MSI more accurate and make our proposed scheme more robust, during the duty period of a slot head, for each active link, the slot head periodically sends a polling message at the broadcast phase. Upon reception of the message, the link receiver will update to the slot head the location information, MSI information, link condition, etc., at the request phase of the same slot

[1]In order for a sender to know the receiving code of the receiver, there should be code assignment/distribution and paging mechanisms. In this research we assume such mechanisms are in place as we mainly focus on call admission and resource allocation.

[2]Before re-sending the request, the call sender should listen to the broadcast message of the slot. If the slot head is changed (which means a new link is admitted into the slot and MSI information is updated), the call sender will choose a new target slot.

in the next frame using the assigned code (in the polling message). In the control message exchange procedure, the slot head may need to send to its own receiver and receive the requests simultaneously. A receiver may also need to receive information transmission from its sender and send update information to the slot head(s) simultaneously. As suggested in [14], full-duplex can be achieved in TH-UWB by blanking the reception at a node during its pulse transmissions. An alternative approach is to use an extra control slot for each frame, which is exclusively used for control message exchanges and information updates. In this case, full-duplex is not required, but the system efficiency will be degraded due to the overhead of the control slot.

Our scheme does not require that each node be able to hear all transmissions from other nodes. Rather, it is required that the broadcast message be heard by all the nodes, and the request and the confirmation be heard by the slot head and the requester, respectively. To achieve this in a short-range UWB network, a solution is to use higher transmission power or more powerful channel codes for these transmissions.

V. Admission Control Algorithm

Upon a call request (for a new link) arrival, the sender first listens to the broadcast channel at each slot, and determines whether or not the call can be admitted and, if admitted, how the resources should be allocated to it (e.g., to which slots at what power/rate levels). We propose to perform the admission and resource allocation at the call sender, instead of the slot heads. Thus the computation complexity can be distributed to the call senders when the calls are admitted, so that the slot heads are not responsible for all the computations.

The relatively long acquisition time in UWB transmissions may significantly limit the resource allocation performance [6]. Therefore, it is critical to keep the effect of acquisition overhead as low as possible, in order to fully explore the high rate transmission. For UWB networks, if a link is scheduled to transmit in two consecutive slots, the acquisition overhead for the second slot is not necessary since the receiver and sender are already synchronized at the first slot. Thus, consecutive slot assignment should be favored.

Fig. 3 shows the procedure of the proposed admission control algorithm, where $\Omega_i \subseteq \{1, 2, ..., M\}$ denotes the set of slots (in each frame) from which link i has gained services. The procedure is detailed as follows. Consider a call request arrives for a new link i with an SINR requirement γ_i and a rate requirement R_i^t. Set $\Omega_i = \phi$ (null set) as link i has not gained any service from any slot. The sender of link i first checks whether there is an *idle slot* (a slot without active links). If so, it chooses the first available idle slot k and, at the same time, acts as the head for the slot. When there is no idle slot available, the call sender determines whether or not the call can be admitted into an already occupied slot. If not, the call is dropped; otherwise, the sender determines in which slot it should transmit (using criterion discussed in Section VI), and sends a request in the request phase of the slot (say k) in the first available frame. If the request is not received correctly (i.e., no confirmation is returned), the sender sends the request in the subsequent frames with a probability

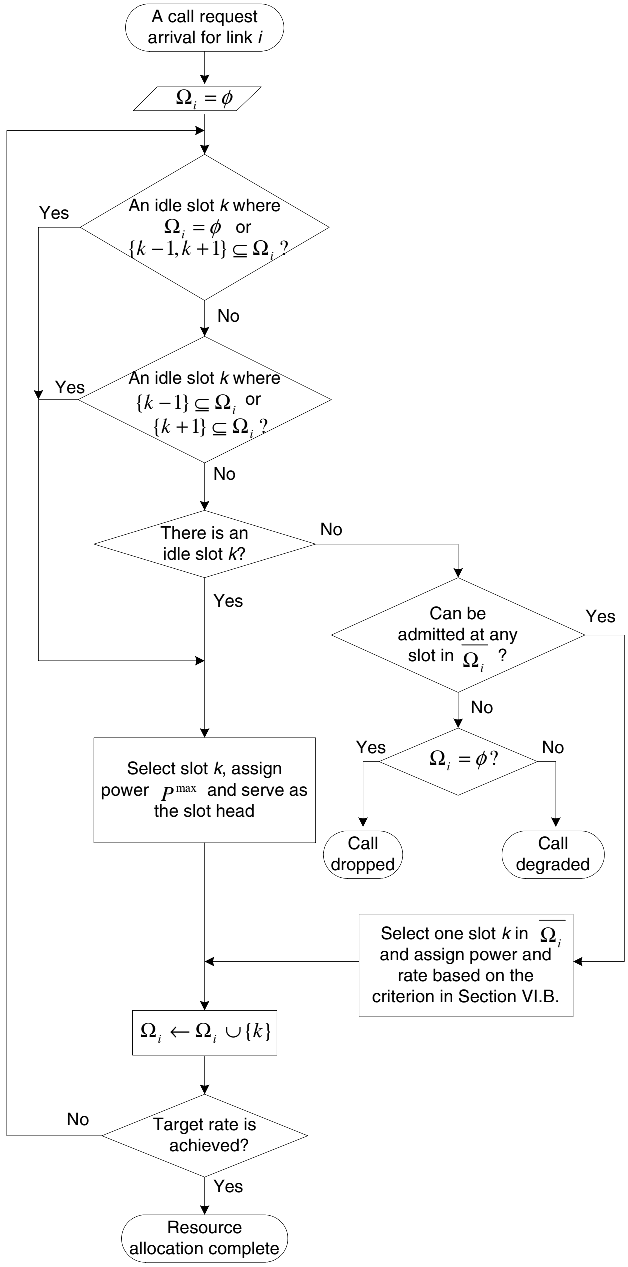

Fig. 3. Procedure of the proposed admission control algorithm.

p until a confirmation from the slot head is received. Then link i joins slot k, and $\Omega_i = \{k\}$.

For each link in a slot, the maximum achieved transmission rate should not exceed $R^{\max} = 1/T_f$. When the admission at a slot cannot meet the rate requirement of link i, the sender will

- check whether there exists an idle slot k (for simplicity, we use the same index k here) whose two neighboring slots are serving link i, i.e., $\{k-1, k+1\} \subseteq \Omega_i$. If yes, choose the slot k; otherwise, go to next step;
- check whether an idle slot k exists with one of its neighboring slots serving link i, i.e., $\{k-1\} \subseteq \Omega_i$ or $\{k+1\} \subseteq \Omega_i$. If yes, choose the slot k; otherwise, go to next step;
- check whether an idle slot k exists. If yes, choose the slot k; otherwise, go to next step;

- check whether or not the call can be admitted into a (non-idle) slot from $\overline{\Omega}_i$. If not, the admission procedure is terminated, and the call is *degraded* since the services that it has received from previous slot allocation cannot meet its rate requirement; otherwise, the call sender selects a slot from $\overline{\Omega}_i$ (based on the criterion in the Section VI-B) and sends a request in the request phase.

For a degraded call, the sender may choose to terminate it if its total achieved rate is less than a threshold αR_i^t $(0 < \alpha < 1)$, and we say that the call is dropped.

The procedure is repeated until the target rate requirement is met, or the call is degraded or dropped.

In summary, for a call request, there are three possible outcomes: dropped, admitted with QoS satisfaction, and degraded with a certain level of QoS satisfaction.

VI. Resource Allocation

A. Resource Definition

Defining the "resource" is a challenging task for UWB networks with peer-to-peer connections. For traditional networks, such as the CDMA cellular networks, the resources can be defined in terms of the *effective bandwidth* [15] for uplink transmission. However, it only applies to the multiple-sender one-receiver case in CDMA cellular networks, and is not suitable for the multiple-sender multiple-receiver UWB networks. Because the interference environment at each receiver may be quite different for peer-to-peer connections, it may not be feasible to find a global resource definition for the whole UWB network. In the following, we define the resource from the viewpoint of each receiver based on the concept of MSI.

From the MSI_i given in (5), it can be seen that when R_i takes the value of the maximum achievable rate given by (3) (if feasible), the MSI_i value is equal to 0. Link i achieves its maximum MSI value when there is no other active link, i.e.,

$$\mathrm{MSI}_i^{\max} = \frac{P_i h_{ii}}{\gamma_i R_i} - \eta_i. \tag{6}$$

For active link i, denote $\Delta\mathrm{MSI}_{ji}$ as its MSI reduction due to the activity of link j, which is given by

$$\Delta\mathrm{MSI}_{ji} = T_f \sigma^2 P_j h_{ji}. \tag{7}$$

Therefore, the MSI of link i can be rewritten as

$$\mathrm{MSI}_i = \mathrm{MSI}_i^{\max} - \sum_{j=1, j\neq i}^{N} \Delta\mathrm{MSI}_{ji}. \tag{8}$$

From (8), to guarantee the transmission accuracy of link i (i.e., a nonnegative MSI_i), the summation of all the $\Delta\mathrm{MSI}_{ji}$ over j should be bounded. From the linear constraint, we define the amount of resources for link i as the $\mathrm{MSI}_i^{\max}$. The $\Delta\mathrm{MSI}_{ji}$ in (8) represents the amount of resources consumed by link j from the perspective of link i, and the MSI_i is the amount of available resources at link i. Such definition can be justified as follows:

- The $\mathrm{MSI}_i^{\max}$ reflects the capability of link i in interference tolerance. With a larger $\mathrm{MSI}_i^{\max}$, more other transmissions can be allowed from link i's viewpoint.
- The spread spectrum nature of UWB transmissions determines that it is desired to incorporate interference into resource definition. This is similar to the uplink of a CDMA cellular network, where the effective bandwidth [15] reflects the interference generated by a user to the reception of other users' signals at the base station.
- Our resource definition is suitable for the peer-to-peer transmissions in the UWB wireless network. If link j's transmitter is far away from link i's receiver, the $\Delta\mathrm{MSI}_{ji}$ is small according to (7), which is consistent with the general principle that two links can be active simultaneously if they are spatially separated far enough.

B. Slot Selection and Power/Rate Allocation

This subsection describes the criterion regarding how a link determines from which slots it requests service and with what power and rate levels it transmits, in order to meet its rate requirement and not to degrade the QoS of existing links.

Consider link i with target rate R_i^t. Let R_i^k denote the transmission rate of link i at slot k, where $k \in \Omega_i$. Then the effective rate of link i, i.e., the total achieved rate seen by the user of link i, is given by

$$R_i^e = \sum_{k\in\Omega_i} R_i^k \cdot \left(1 - I_{\overline{\Omega}_i}(k-1)\cdot\xi\right) \cdot \frac{T_{\text{slot}}}{T_{\text{frame}}} \tag{9}$$

where T_{frame} and T_{slot} are the length of a frame and a time slot as shown in Fig. 2, respectively, ξ is the acquisition overhead ratio (i.e., the fraction of time for acquisition in an slot if acquisition is needed), and the indication function

$$I_{\overline{\Omega}_i}(k-1) = \begin{cases} 1, & k-1 \in \overline{\Omega}_i \text{ or } k-1 = 0 \\ 0, & \text{otherwise.} \end{cases} \tag{10}$$

When slot $k-1$ is serving link i, the re-acquisition overhead in slot k is not necessary.

When a new call request for link i arrives, by checking whether there exists at least one idle slot, the sender node selects the first idle slot k and transmits with maximum power $P_i^k = P^{\max}$ and rate R_i^k so as to achieve the target R_i^t:

$$R_i^k: \quad R_i^t = R_i^k \cdot (1-\xi) \cdot \frac{T_{\text{slot}}}{T_{\text{frame}}} \tag{11}$$

i.e., $R_i^k = R_i^t/[(1-\xi)\cdot\frac{T_{\text{slot}}}{T_{\text{frame}}}]$. If $R_i^t/[(1-\xi)\cdot\frac{T_{\text{slot}}}{T_{\text{frame}}}] > R^{\max}$, we set $R_i^k = R^{\max}$. This means, when the target rate R_i^t cannot be achieved through slot k, we set R_i^k to be the maximum achievable rate $R^{\max}$ at a slot.

If no idle slot exists, the call sender of link i selects one among all the M slots. For the slot selection, let Q_i^k denote a penalty function of the resource consumption by link i at slot k, U_i^k (a utility function) the gain of link i in its effective rate, and C_i^k (a cost function) the overall penalty versus utility for link i at slot k. To achieve efficient resource utilization, the slot with the minimal cost function is chosen, as discussed in the following.

Let $N(k)$ denote the number of existing links within slot $k \in \{1, ..., M\}$. Let P_j^k and R_j^k denote the transmission power and rate, respectively, for an active link j at slot k. The MSI value (i.e., the amount of available resources) of active link j at slot k is given by

$$\mathrm{MSI}_j^k = \frac{P_j^k h_{jj}}{\gamma_j R_j^k} - \eta_j - T_f\sigma^2 \sum_{l=1, l\neq j}^{N(k)} P_l^k h_{lj}. \tag{12}$$

Under the hypothesis that the new call request (for link i) selects slot k for service, let p_i^k and r_i^k denote the power and rate of link i at slot k, respectively[3]. The power p_i^k should be constrained by the MSIs of existing active links at slot k [3], i.e.,

$$p_i^k = \min\big\{P^{\max},\ \min_{1\le j\le N(k)}\{\frac{\mathrm{MSI}_j^k}{T_f\sigma^2 h_{ij}}\}\big\}. \quad (13)$$

If $p_i^k = 0$, link i cannot be admitted at slot k as it will violate other links' MSIs. Thus we set the cost function of link i at slot k as $C_i^k = \infty$; If $p_i^k > 0$, we set r_i^k so as to achieve the target rate R_i^t, i.e.,

$$r_i^k: \quad R_i^t = r_i^k \cdot (1-\xi)\cdot \frac{T_{\text{slot}}}{T_{\text{frame}}} \quad (14)$$

where r_i^k is bounded by $R^{\max}$. This setting is feasible only if the resulting MSI of link i at slot k is nonnegative, i.e.,

$$\frac{p_i^k h_{ii}}{\gamma_i r_i^k} - \eta_i - T_f\sigma^2 \sum_{j=1}^{N(k)} P_j^k h_{ji} \ge 0. \quad (15)$$

If (15) cannot be satisfied (i.e., the SINR of link i at slot k is not high enough to support r_i^k), we set $C_i^k = \infty$.[4]

Under the hypothesis that link i is admitted in slot k, the MSI reduction of an existing active link j ($1 \le j \le N(k)$) at slot k due to the admission of link i (or equivalently, the amount of link j's available resources which will be consumed by link i) is given by $\Delta\mathrm{MSI}_{ij}^k = T_f\sigma^2 p_i^k h_{ij}$.

The penalty function Q_i^k should reflect the resource consumption by link i if it is admitted at slot k. One intuitive way to define Q_i^k is to aggregate all the $\Delta\mathrm{MSI}_{ij}^k$, $1 \le j \le N(k)$. However, it is also desired to differentiate existing links at slot k with different MSI_j^k values. In general, with the same value of $\Delta\mathrm{MSI}_{ij}^k$, the MSI reduction of an existing link j with a small MSI_j^k should cause large penalty as the link j may become the bottleneck for subsequent new link admissions. Therefore, the penalty induced by admitting link i at slot k is defined by

$$Q_i^k = \sum_{j=1}^{N(k)} \frac{\Delta\mathrm{MSI}_{ij}^k}{\mathrm{MSI}_j^k}. \quad (16)$$

To reflect the increase of effective rate achieved from link i being active in slot k, the utility function is defined by

$$U_i^k = r_i^k \cdot (1-\xi)\cdot \frac{T_{\text{slot}}}{T_{\text{frame}}}. \quad (17)$$

We use a heuristic cost function C_i^k to deal with the overall penalty versus gained effective rate, i.e., $C_i^k = Q_i^k / U_i^k$.

For the call request of new link i, if all the cost functions at the M slots are infinity, the call will be dropped; otherwise, the target slot k^* is chosen by

$$k^* = \arg\min_{1\le k\le M} C_i^k \quad (18)$$

with the power level $P_i^{k^*} = p_i^{k^*}$ and rate level $R_i^{k^*} = r_i^{k^*}$. Once the target slot k^* is chosen, the source node sends a request at the request phase of slot k^* and expects to receive a confirmation. After that, $\Omega_i = \{k^*\}$.

It is possible that the effective rate of link i in Ω_i calculated by (9) is less than its rate requirement R_i^t. In such circumstances, the call sender needs to request services at more slots (not in Ω_i) until the rate requirement of the call is satisfied or the call is degraded. If there is no idle slot[5], a procedure similar to that described above is executed, except for the determination of r_i^k and the utility calculation. Combined with previous allocated rates in other slots (in Ω_i), r_i^k is given by the value that satisfies the target rate requirement:

$$r_i^k: \sum_{l\in\Omega_i} R_i^l \cdot \big(1 - I_{\overline{\Omega}_i}(l-1)\cdot\xi\big)\cdot\frac{T_{\text{slot}}}{T_{\text{frame}}} + r_i^k\cdot\big(1-I_{\overline{\Omega}_i}(k-1)\cdot\xi\big)\cdot\frac{T_{\text{slot}}}{T_{\text{frame}}} + R_i^{k+1}\cdot I_{\Omega_i}(k+1)\cdot\xi\cdot\frac{T_{\text{slot}}}{T_{\text{frame}}} = R_i^t. \quad (19)$$

Similarly, if $r_i^k > R^{\max}$, we set $r_i^k = R^{\max}$. The second term on the left side of (19) means the effective rate at slot k, and the third term represents the increase of effective rate at slot $k+1$ if it already provides service to link i (because of the reduction of the acquisition overhead at slot $k+1$). Accordingly, the utility is given by

$$U_i^k = r_i^k\cdot\big(1-I_{\overline{\Omega}_i}(k-1)\cdot\xi\big)\cdot\frac{T_{\text{slot}}}{T_{\text{frame}}} + R_i^{k+1}\cdot I_{\Omega_i}(k+1)\cdot\xi\cdot\frac{T_{\text{slot}}}{T_{\text{frame}}}.$$

In (19), it is possible that the calculated $r_i^k \le 0$. This happens when the target rate can be met by the increase of effective rate at slot $k+1$, i.e.,

$$\sum_{l\in\Omega_i} R_i^l \cdot \big(1 - I_{\overline{\Omega}_i}(l-1)\cdot\xi\big)\cdot\frac{T_{\text{slot}}}{T_{\text{frame}}} + R_i^{k+1}\cdot I_{\Omega_i}(k+1)\cdot\xi\cdot\frac{T_{\text{slot}}}{T_{\text{frame}}} \ge R_i^t. \quad (20)$$

In this case, the assigned r_i^k should have a small but positive value to maintain the link activity at slot k.

The above procedure continues until the rate requirement of the call for link i is met or the call is degraded.

VII. Performance Evaluation

In this section, we present simulation results to evaluate the performance of the proposed scheme and compare it with the MSI-based scheme in [3]. The experimental network is set up as follows. A number of stationary nodes are uniformly distributed in a two-dimensional 100 m $\times$ 100 m square. The call arrival to the whole network is a Poisson process with rate λ, and each call duration is exponentially distributed with mean value equal to μ. For the simplicity of presentation, each call arrival is assigned a sender and a receiver, both independently and uniformly located in the 100 m $\times$ 100 m square (thus referred to as *uniform topology*). The time frame structure, as illustrated in Fig. 2, has a fixed period of 30 ms, which is further divided into a beacon (with duration 5 ms)

[3]We use capital letters "P" and "R" to denote the actual power and rate values already allocated at a slot, and use the lower case letters "p" and "r" to denote those in hypothesis.

[4]One argument is that, if the SINR of link i at slot k is not high enough, a lower rate is used for link i at slot k. However, we do not adopt this because, if link i is admitted at slot k using a lower rate, its MSI value at this slot may be very low, thus making it a bottleneck in subsequent resource allocation of other new links at slot k.

[5]If there exists at least an idle slot, the procedure discussed in Section V is used.

TABLE II
PARAMETERS USED IN THE PERFORMANCE EVALUATION

Parameters	Values
θ	2.4
K	1
η	2.568×10^{-21} W/Hz
γ	7 dB
T_f	100 ns
σ^2	1.9966×10^{-3}
$P^{\max}$	0.5 mW
R_i^t	2 Mbps
α	0.1
$\{p_1, p_2, .., p_m\}$	{0.1, 0.2, 0.4, 0.8}

and 5 time slots (each with duration 5 ms). For comparison, a similar time frame structure is implemented in the MSI-based scheme except that there is only one time slot (with duration 25 ms) in a frame. Our proposed control message exchange procedure is also used in the MSI-based scheme. In addition, the acquisition time in both schemes is the same, which leads to the acquisition overhead ratio in our scheme 5 times that in the MSI-based scheme. For presentation simplicity, the acquisition time in both schemes is measured by the acquisition overhead ratio ξ in our scheme. Other relevant parameters used in the simulation are given in Table II. All simulation results are obtained by averaging over 2,000 calls.

A. Throughput

The achieved average overall throughput versus the call arrival rate λ is shown in Fig. 4 with $\xi = 0.5$ and $\mu = 60$ s. It is observed that our scheme achieves up to 70% increase in the system throughput over that of the MSI-based scheme. The gain becomes more significant as the call arrival rate λ increases. This is because our scheme can solve the near-sender-blocking problem by temporal exclusion. Therefore, the system employing our scheme can admit more calls. Although the use of multiple slots in our scheme may have the risk of aggravating the overhead due to acquisition, the operation explained in Sections V and VI tends to regulate each active node acquiring consecutive slots and thus reduces the impacts of long acquisition time.

B. Call Dropping Probability

To measure the performance of our admission algorithm, we observe the successful call admission. A call is dropped if its admission may damage the reception of existing calls, or equivalently, there are no sufficient resources available at the instant that the sender requests for an admission. Here we assume there is no retry, which may be added to the scheme for implementation purposes. Fig. 5 shows that our scheme provides a relatively small likelihood of call dropping than the MSI-based scheme. It is further observed that some calls may be admitted but unable to acquire sufficient resources to satisfy their QoS requirements (i.e., degraded). As the offered load increases, more calls are likely to be degraded. Note that there is no degraded call in the MSI-based scheme (with only one slot in each frame) to avoid bottleneck effect.

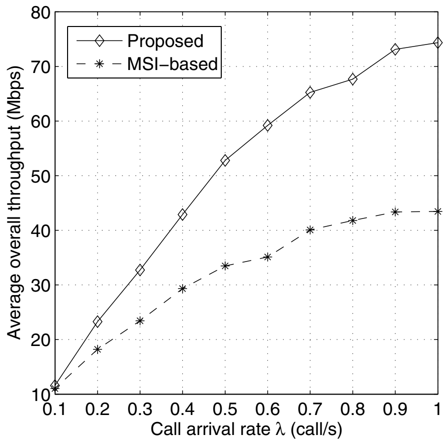

Fig. 4. Throughput over uniform topology.

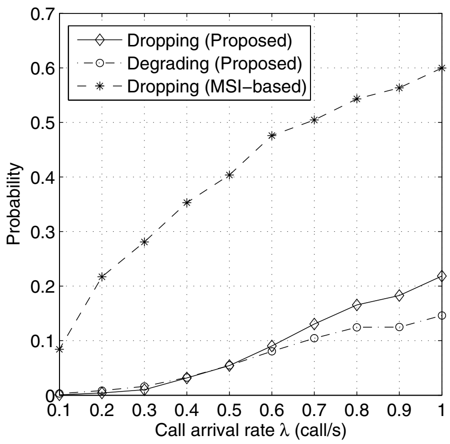

Fig. 5. Probability of call being dropped or degraded over uniform topology.

C. Power Consumption

Besides the throughput and call dropping probability, another concern of the resource management in the UWB network is the power consumption. Fig. 6 shows the average transmission power consumption of each admitted call. Here we ignore the energy consumed in sending the control/update packets in our scheme and the MSI-based scheme. It can be seen that our scheme can reach a higher throughput with much lower average power consumption. Furthermore, when the traffic load increases, the average power consumption slightly decreases. This can be explained as follows. When the call arrival rate is small, almost all the existing active links at each slot have sufficient MSI values. Thus a new link is very likely to transmit with maximum power $P^{\max}$ at its serving slots. When the call arrival rate increases (after the threshold $\lambda = 0.5$ call/s in the example), there are more active links at a slot, resulting in insufficient MSI values of some links. It is likely that the senders of the new calls are not allowed to transmit with $P^{\max}$ due to the constraints of MSIs of existing active links. Thus, the average power consumption slightly decreases.

TABLE III
CAPACITY COMPARISON OF UNIFORM AND CLUSTERED TOPOLOGIES

Topology	λ (call/s)	Throughput (Mbps)	Dropping Prob. (%)	Degrading Prob. (%)	Average link number per slot
Uniform	2.0	92	43	16	18
	1.5	84	37	14	17
	1.0	75	22	14	17
Clustered	2.0	170	16	11	32
	1.5	156	6	7	31
	1.0	118	1	2	25

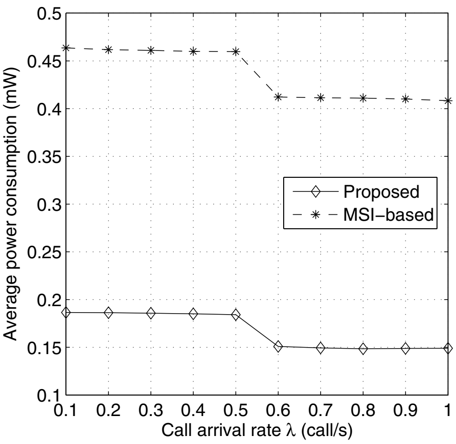

Fig. 6. Average power consumption over uniform topology.

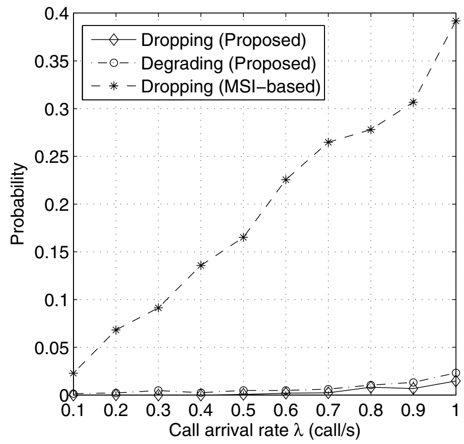

Fig. 8. Probability of call being dropped or degraded over clustered topology.

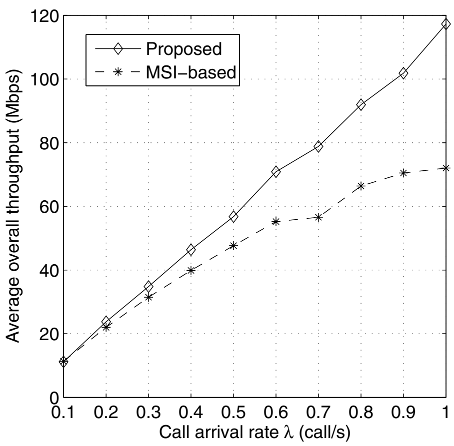

Fig. 7. Throughput over clustered topology.

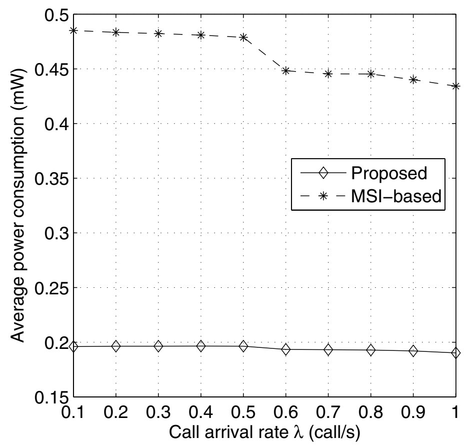

Fig. 9. Average power consumption over clustered topology.

D. Performance in a Clustered UWB Network

Consider a clustered UWB network where the 100 m $\times$ 100 m square is equally partitioned into four regions, and each node only communicates with nodes in the same region. A node can hear any other node's transmission as long as it tunes to the transmission code and the received SINR exceeds a threshold. The beacon and broadcast messages can be heard by all nodes, the request can be heard by the slot head, and the confirmation can be heard by the requester. Call arrivals are distributed in different regions so that calls are equally located in the experimental area. Figs. 7-9 illustrate the behavior of the proposed scheme as compared to the MSI-based scheme over the clustered topology. Similar to the result of the uniform topology, Fig. 7 shows that both schemes achieve nearly the same throughput when the traffic density is low. As the call arrival rate increases, the throughput improvement achieved by our scheme is more significant. In terms of the throughput and call dropping/degrading probability, it can be seen from Figs. 4, 5, 7, and 8 that both schemes perform better in the

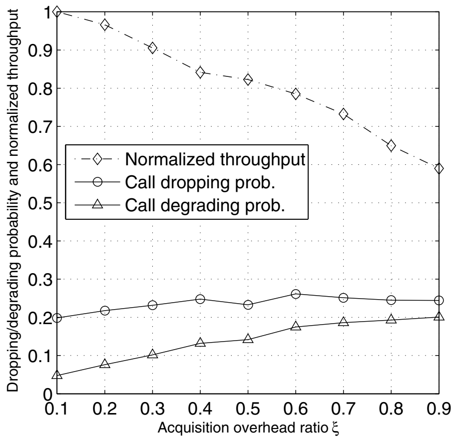

Fig. 10. Call dropping/degrading probability and normalized throughput (with respect to $\xi = 0.1$ case) versus acquisition overhead ratio ξ in our proposed scheme.

clustered topology than in the uniform topology. The power consumption shown in Fig. 9 reveals the similar tendency to that of the uniform topology.

To further compare the network capacity with the uniform and clustered topologies, Table III shows the system throughput, dropping and degrading probabilities, and average number of active links per slot in our scheme with different topologies. It can be seen that, with the uniform topology, $\lambda = 1$ call/s makes the network saturated with dropping probability 22%. When λ increases to 2 call/s, the throughput increase is relatively small. However, with the clustered topology, $\lambda = 1.5$ call/s makes the network close to saturation, with dropping probability 6%. This is because the path gain of each link in the clustered topology is likely larger than that in the uniform topology, which in turn increases the MSI of each link. Thus, more links can be admitted in a slot with the clustered topology than with the uniform topology, as shown in Table III. This implies that routing at the network layer should be jointly designed with the radio resource allocation at the link layer. For a link with the sender and receiver separated far away, a single hop transmission may lead to a small delay, at the cost of the possibility of its becoming a bottleneck due to the small MSI, while multi-hop transmissions can keep a large MSI for each hop, at the cost of a large delay due to the multi-hop link. How to achieve an appropriate tradeoff is an interesting issue for further research.

E. Effects of Acquisition Overhead

The impacts of acquisition overhead on the overall network performance such as call dropping and degrading probabilities and normalized throughput (with respect to $\xi = 0.1$) are shown in Fig. 10 for our proposed scheme with $\lambda = 1$ call/s. As expected, a higher acquisition overhead ratio tends to reduce the achievable throughput. On the other hand, if a transmission encounters a higher acquisition overhead ratio, the effective transmission time is reduced, which leads to an increasing number of calls without transmission rate requirement satisfaction. This illustrates the increase of call degrading probability as the acquisition overhead ratio increases. From Fig. 10 it can be seen that, when the acquisition overhead ratio increases from 0.1 to 0.9 (i.e., the transmission efficiency significantly decreases from 90% to 10% when acquisition is needed), the average overall throughput only decreases by approximately 40%. This is because our proposed scheme favors transmissions of a link at consecutive slots.

VIII. CONCLUSION

We have proposed a resource management scheme for short-range UWB wireless networks. The scheme can address the near-sender-blocking problem by a temporal exclusion mechanism. With the aid from the rotated slot heads, the information exchange among the nodes can be performed effectively, and the computation complexity and power consumption of slot heads can be fully distributed to the call senders when the calls are admitted into one or more slots. In addition, the admission control and resource allocation algorithms can be executed at the call senders upon the call arrivals. The slot selection criterion favors consecutive slot assignment, thus alleviating the negative effect of long acquisition time in UWB transmissions. Further research issues include joint routing and resource management, and scalability in large UWB networks.

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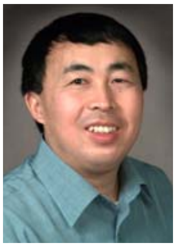

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