A New Utility Based Channel Allocation Scheme for P2P Enabled TDD CDMA System

Yan Zhang
Department of Electronic Engineering
Shanghai Jiao Tong University
Shanghai, PRC
Email: zyan@sjtu.edu.cn

Dan Shang
Wireless Communication Lab
Philips Research East Asia
Shanghai, PRC
Email: dan.shang@philips.com

***Abstract*—Peer-to-peer (P2P) communication in TDD CDMA networks is a new method where nearby users can direct communicate with each other without the being relayed by base stations. It can save nearly half radio resource and significantly improve the capacity for the applications where communication participators are close to each other. An appropriate RRM (radio resource management) mechanism is the key component to realize this new system. In this paper, we focus on the channel allocation aspect of RRM, providing a new scheme to coordinate between the hybrid traffic in P2P enabled system. Analytical and simulation results show that the utility-based scheme achieves much lower packet delay with the same throughput, comparing with conventional methods. Fairness is also taken into account by using waiting time as a parameter to determine the transmission priority, and hence the benefits of users with poor link conditions are guaranteed.**

Keywords- weight-based, channel, allocation, P2P, CDMA, system capacity, fairness

I. INTRODUCTION

Under a conventional cellular architecture, one mobile terminal has to communicate with another terminal via the relay of the base station even when the two terminals are very close to each other. It is obviously more efficient in account of radio resource and power consumption if they can directly communicate without base station's relay. This is called P2P direct communication in cellular networks, illustrated as Figure 1. It is a primitive form of a self-organizing network (e.g. ad-hoc network), where terminals communicate directly without any pre-existing infrastructure.

As shown in Figure 1, the dash line represents the network control information between P2P terminals and the base station. The dash line also indicates that the information is only exchanged occasionally. Data traffic is transmitted from one terminal to another directly. Without the relay of the base station, P2P direct connection can significantly expands the system capacity. For a terminal, this communication method can also save transmission power.

Because P2P communication occurs under the control of mobile network infrastructure in this mixed architecture, the complexity due to self-RRM at the terminal end is greatly reduced. Furthermore, since the mobile transmits and receives data at the same frequency in a TDD CDMA system, the modification can be kept at a minimum level to realize the P2P enabled system based on a traditional cellular system. Thus P2P application is very promising in TDD CDMA system.

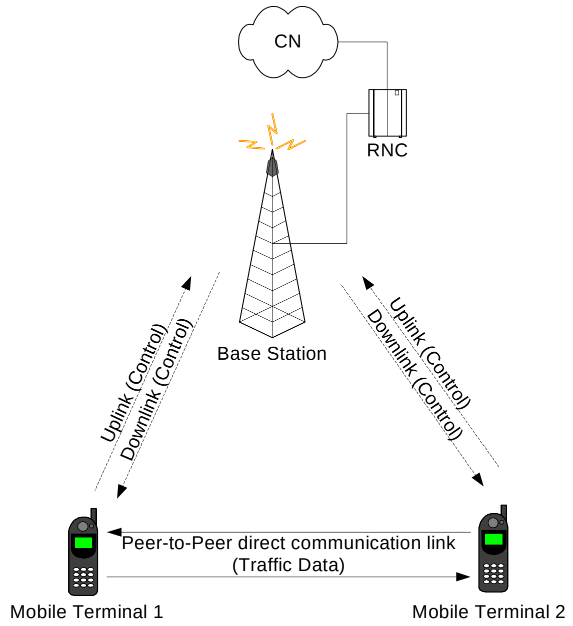


Figure 1. P2P Communication Architecture

However, to embed the new application to an existing system, modifications in RRM strategy is unavoidable as the management objective and environment has changed. Although many well-designed RRM algorithms based on different philosophies have achieved good performance [2-5] in conventional cellular systems, none of the existing solutions can be directly applied to the P2P enabled system. Firstly, network cannot obtain actual information about the resource requirement and the changing communication environment of P2P users. Secondly, the idea of strictly defined uplink and downlink timeslots is blurred in a P2P system. A P2P user may transmit data in a downlink timeslot or receive in an uplink timeslot, resulting in additional interference to the existing system. These issues are unique to a P2P enabled system, and no previous strategies once take them into consideration. Therefore the channel allocation strategy needs to be modified to adapt to the changes in system architecture.

To address the problems above, we develop a utility-based channel allocation strategy to improve the performance of a P2P enabled TDD-CDMA system in a mixed environment with conventional systems. The resource is competed between conventional and P2P users dynamically in the light of calculated utility. As for the balance in choosing conventional or direct link is not

The rest of this paper is organized as follows. In section II, we introduce the network, traffic, path loss and queue model used in simulation. Afterwards, utility-based channel allocation scheme is presented with details. The simulation results and evolutions are shown in section IV, while conclusion is given in the end.

II. System Model

A. Network Model

The cellular network model used in our simulation consists twelve cells with wrap up technique to reduce margin effects. The cells are distributed as the diagram shown in Figure 2:

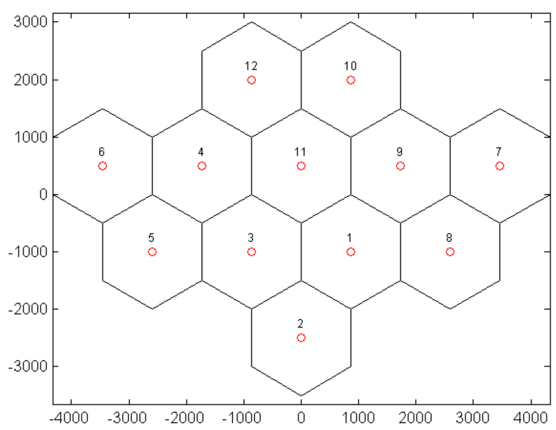


Figure 2. layout of cellular architecture (Both x-axis and y-axis are in the unit of meter.)

Base stations are situated at the center of each cell. Users are uniformly distributed in each cell.

B. Traffic Model

With the increasing prevalence of wireless communication, all the typical services in a wired system such as voice, data and multimedia are expected to be provided in a wireless communication system. Some of them are already quite popular nowadays.

To accurately depict the hybrid traffic condition, we use different models to describe different services respectively. For voice service, a Poisson arrival process gives a pertinent description to the stream characteristics, and we continue to use in our simulation. While for data and multimedia services, due to their diversities, no mature model exists. Typically the WWW data model [6] is used for reference to depict Internet service, in which the traffic is routed from the server to the clients. As a result, this model is not consistent with current applications in wireless systems, especially in the case of increasing uploading service. Here we would like to apply a simplified data model with the same traffic loads in both link directions in our simulation. In the new model, the packets of data service are generated in a process with a geometrical distribution as described in [6].

Traffic from different service is generated independently with respective distributions. All the traffic streams can be routed from one user to the other either directly for P2P users or indirectly (via base station) for conventional users.

C. Pathloss Model

In this simulation platform, omni directional antenna is applied. For the propagation model, it's a little complicated for a P2P enabled TD-SCDMA system. Different from other kinds of wireless system, three kinds of propagation models in outdoor-to-indoor and pedestrian test environment are included. They are the propagation models between BS&MS, MS&MS and BS&BS, which are expressed by following equations.

- Between BS and MS

$$L_{outdoor-to-indoor,BS-MS} = 30Log_{10}(f) + 40Log_{10}(R) + 49$$

- Between MS and MS

$$L_{outdoor-to-indoor,MS-BS} = 30Log_{10}(f) + 40Log_{10}(R) + 49 + \Delta_1 \quad \Delta_1 > 0$$

- Between BS and BS

$$L_{outdoor-to-indoor} = 30Log_{10}(f) + 40Log_{10}(R) + 49 + \Delta_2 \quad \Delta_2 < 0$$

Δ_1 and Δ_2 are the compensatory values that can be alternated. They are considered because the alternation of antenna height compared with the conventional radio system. It is well known that in conventional cellular system the radio communication link exists only between the mobile terminal and a base station with antenna located in a relative high position. However, in P2P communication, the mobile terminals are always placed in briefcase or pockets. That means both transmitter and receiver antennas of the two communicating mobile terminals are likely to be closer to the ground. The antennas held adjacent to head also causes significant cross-polarization of the received signal waveform. Therefore, Δ_1 in equation (2) should be carefully decided. In the simulation, we use +5dB and –5dB for Δ_1 and Δ_2 respectively.

D. Queue Model

In TDD CDMA system, power control process will regulate the users in the same timeslot with the most stringent BER requirement. To avoid high power consumption and even lower system capacity due to traffics with different BER requirements sharing a single timeslot, we separate traffic in queues on account of their particular BER requirements. Thus, in the simulation, arriving services are put in different queues according to their BER, and it is devised to try to assign streams from the same queue to a timeslot.

In this model, BER is used as the gauge to divide multiple queues. The ID of queues for a certain stream is given by

$$i = -\log(BER/10^{-3}) \qquad (1)$$

where 10^{-3} represent the typical the BER requirement in voice service. For example, if a stream with $BER = 10^{-5}$, it is assigned to queue 2.

The separation of queues only give the preference in assigning timeslots, while no priority is decided between queues. Although real-time traffic always has highest priority, resource is competed dynamically by all the queues in the light of calculated utility value as described afterward, and there are no predefined partitions of timeslots among queues. However, if a user from a queue occupies a timeslot first, this timeslot only serves the users from the same queue, except that there is no other resource available.

III. Utility Based Channel Allocation Strategy

Previous approaches require instant link and traffic information, which are not available under P2P architecture. In addition, new interference caused in the P2P system makes the channel allocation even complicated. To handle the problems unique to a P2P enabled system as well as the problems common to a wireless channel, a utility-based algorithm is presented in this paper. We quantify all the correlative factors into a single utility value to indicate the priority of a stream for system resource allocation. Three important parameters are taken into account to form the utility function: traffic priority, data delayed time and stream length. Arriving data may have a pre-established priority, which should be respected in resource scheduling. Non-real-time traffic will be timeout after a certain delay tolerance. The longer the stream is, the more resource it will occupy. Thus, it is reasonable to reduce its priority to avoid block other traffic. As for other parameters, such as link quality, are not included. Because it is not easy to get frequently updated values in a P2P enabled system or the contribution to improve the system performance is not comparable to the cost of algorithm complexity.

The utility value $\phi_k{}^i$ for user k in queue i can be calculated as:

$$\phi_k{}^i = \frac{w \cdot \delta_{pri} + \delta_\tau}{\delta_{packetlength}} \qquad (2)$$

where δ_{pri} indicates whether the stream is time-critical traffic, such as voice and multimedia. If it is, δ_{pri} is set to one for real time traffic and zero for others.

$$\delta_\tau = \begin{cases} 0 & time-critical\ traffic \\ \dfrac{1}{\tau} & others \end{cases} \qquad (3)$$

$\tau = \tau_{timeout} - \tau_{wait}$, in which τ_{wait} is the time duration that the traffic has waited already and $\tau_{timeout}$ is the total time period that the traffic can be backlogged. δ_τ is inversely proportional to τ for non-real-time traffic and w is the weight.

$\delta_{packetlength}$ is proportional to the amount of the data traffic. The more traffic a user wants to send, the lower priority it should has. In voice service, $\delta_{packetlength}$ is set to 1. The reason to do so is that voice service is a real time service, and no packets will accumulate at the sender end. For data service, the parameter indicates the amount of packets a user wants to send in each session.

Since synthesized utility is the only reference in resource management, the weight w should be well designed to make a pertinent coordination among users according to the varying traffic and communication environment.

Again, channel allocation to P2P users is also constrained by the interference scenario. $\varphi_{j,k}$ is used to indicate whether timeslot j can be used by P2P user k. If there are no conventional users within a certain distance to k using the same timeslot in opposite direction, it will be assigned as 1.

$$\varphi_{j,k} = \begin{cases} 1 & no\ users_{conv}\ in\ TS_j \\ & in\ opposite\ direction \\ 0 & otherwise \end{cases} \qquad (4)$$

The distance threshold can be set in consideration of path loss model. Since P2P communication is usually constrained to be within a certain distance threshold to guarantee link quality of direct connection, we can estimate the transmission power of the P2P terminal. For conventional users, uplink power control is used to overcome near and far effect. We can easily estimate the maximum transmission power of a conventional user when it is at the edge of the cell. Thus, a distance threshold is reckoned to facilitate getting a satisfactory SIR target at the P2P receiver.

Hence, all the arriving traffics acquire resource in a descending order of their calculated utilities. Traffics from different queues are avoided to camp within the same timeslot. Since P2P receiver gets the data at the same time as the sender transmits, P2P users should be served in pairs. We set a receiver the same weight as the sender's. If only the constraint $\varphi_{j,k}$ is 1 for both of a P2P pair, this timeslot k is allocated to the p2p pair.

IV. Simulation And Results

To evaluate the proposed utility-based resource allocation scheme, a classical Round Robin algorithm is also evaluated as the comparison to the new scheme in a P2P enabled TDD CDMA system.

The simulation is performed in a multi-cell and wrap-around environment. When a user reaches the edge of the network, it continues to the other side of the network instead of bouncing back, which avoids edge effect. Since short distance P2P transmission usually gets better performance, we assume the BER requirement of P2P traffic is 10^{-6} for data service and 10^{-4} for voice service, while the requirement of conventional traffic is 10^{-5} and 10^{-3} respectively. If data streams arriving at the receiver end are under the BER threshold, they should be re-transmitted. δ_{pri} is set to 5 for real-time traffic and 0 for non real time traffic, and hence m equals 5. $\tau_{timeout}$ is set to 50 sub frames here. $\delta_{packetlength}$ is the amount of data generated in each session in data service. Since $\delta_{packetlength}$ is always 1 in voice service, the highest priority 5 is acquired by voice user. The percentage of P2P users ranges from 0 to 1 with 0.1 intervals. The simulation lasts 1.5s (300 sub frames).

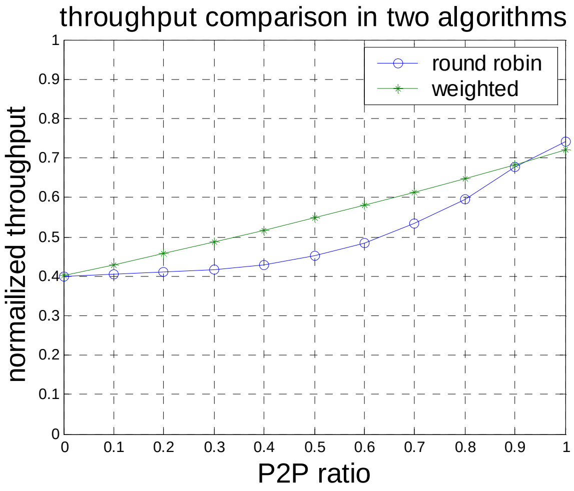


Figure 3. Throughput Comparison

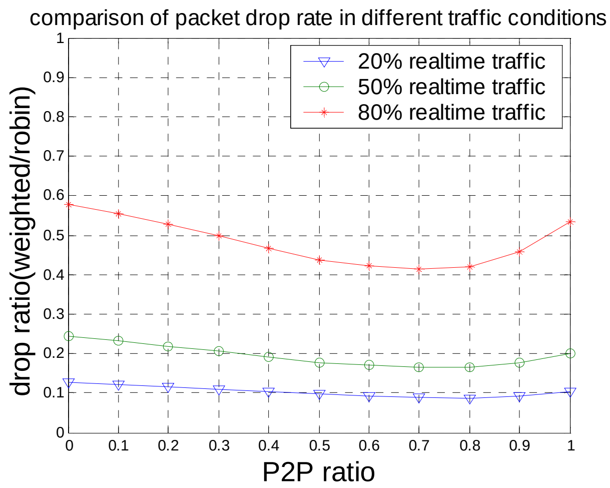
In Figure3, normalized throughput is defined as the actual throughput divided by the maximum possible throughput of the system. In a P2P enabled system, the maximum throughput is acquired under the condition that all the resource is used for P2P communication, and all users are with perfect link (no re-transmission exists). Since control information between P2P users and the base station is only needed at the beginning of a session, we omit the signaling in our simulation. Thus, because conventional communication needs the relaying of the network and occupies twice the radio resource as that of the P2P users, it can be easily assumed that the throughput at P2P ratio 1 should be doubled in comparison to P2P ratio 0. However, the result from figure 2 is not that promising. This is due to the increasing P2P users also introduce more interference to each other, while in a conventional way the base station will always adjust the transmission power of users dynamically and limit strong interference from certain users to others.

Throughput in the two algorithms is comparable, while the curve indicating round robin algorithm is concave. This is due to that conventional users are polled before all the P2P users in round robin, and in another word, they have higher priority and delayed the climbing of the throughput line.

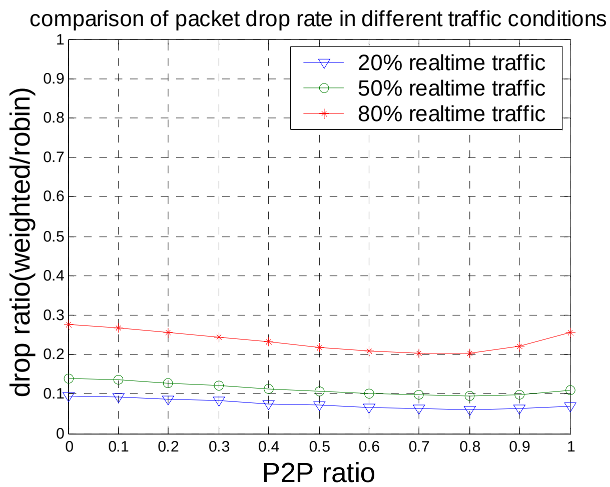
For real-time traffic, a packet is dropped if only it does not get the required resource, while for non-real-time traffic, it will not be dropped until it is time out. Performance on packet drop rate is the main interest we have in simulation.

Due to the big difference between the two algorithms, the ratio of packets dropped in utility-based scheme to those in round robin is used as y-axis in Figure 4.

As shown in the figure, different percentages of real-time traffic are used for comparison in simulation. Since voice service is already a very mature application, the percentage of voice traffic become smaller and smaller with the increasing capacity of a wireless communication system. Presently, due to some technique reasons, multimedia application is not popular yet, and the ratio of real-time traffic to the whole traffic stream decreases to some extends. However, the condition will be changed in the near future. Thus, in order to get an all-around idea of the performance of the new scheme, three typical traffic conditions are simulated. The results show that the performances of the two algorithms are sensitive to traffic conditions.



(a) Heavy Load



(b) Light Load

Figure 4. Comparison of Packet Drop Rate

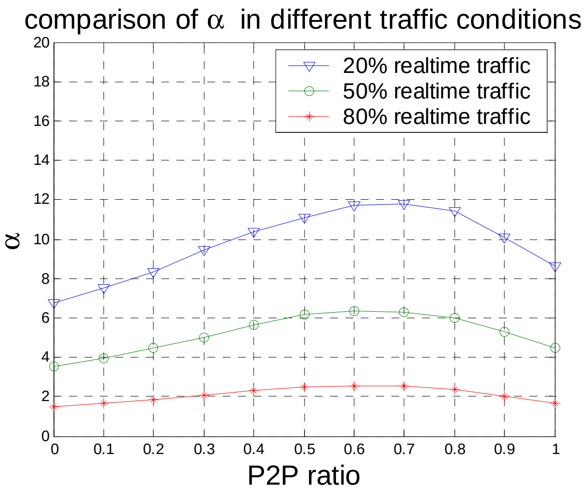
As we set the traffics generate in a speed more than the system's capacity to simulate the ultimate condition, there are always packets being dropped and the sum of dropped packets increases with the up going real-time traffic. By adjusting the traffic generation parameters, we get different traffic load to study the performance. However, even if real time traffic is eighty percent in a heavy-loaded system, the amount of dropped packets in utility-based strategy is still about half of that in round robin. While in the light-loaded system, the performance of weighted scheme is much better than the other.

It can be concluded that our strategy can effectively decrease drop rate, even if real time traffic is very heavy-loaded.

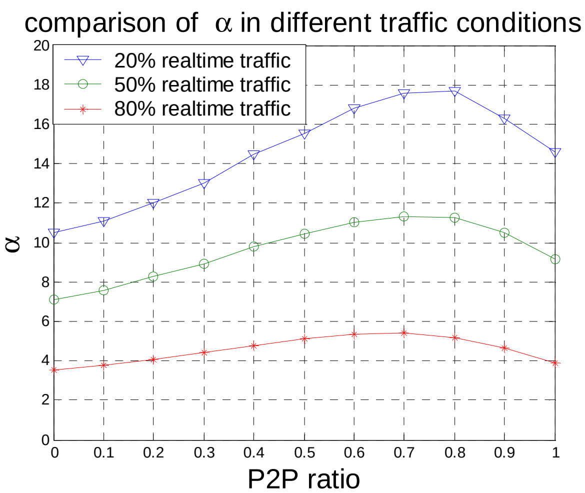
However, drop rate is not a full-scale target to gauge the performance. We cannot tell that a system with high re-transmission rate and low drop rate is better than a system with low re-transmission rate and high drop rate. Thus, to exclude the severe influence of data model and parameters on the result and to get a more disinterested comparison, a new coefficient α is defined as

$$\alpha = \frac{PK_{trans\_utility} / PK_{drop\_utility}}{PK_{trans\_robin} / PK_{drop\_robin}} \qquad (5)$$

where PK_{trans} is the amount of packets transmitted successfully, and PK_{drop} is the amount of packets dropped. The ratio indicates the difference between the capabilities of two algorithms to schedule traffic.



(a) Heavy Load



(b) Light Load

Figure 5. Comparison of α

Since the more the packets transmitted and the less the packets dropped in the utility based scheme than round robin method, the larger the α is. Thus, α indicates the difference on the transmission performance of the two schemes. From the comparison of (a) and (b) in Figure 5, the performance of our algorithm is obviously superior to round robin by several folds under different traffic-load conditions. Only when the heavy-load traffic consists mostly of real time traffic, there is no very appealing performance improvement with the new algorithm.

V. CONCLUSION

In this paper, a time-varying utility-based channel allocation algorithm is proposed and evaluated by simulations. Round robin algorithm is also investigated in the simulation for comparison purpose. With different traffic load and percentage of real time traffic, the new algorithm outperforms round robin in different degrees. When the traffic is extremely heavy-loaded, the advantage of utility-based scheme is slightly weakened.

By examining time varying traffic and communication environment parameters, the new algorithm achieves quite outstanding performance at a very low cost in terms of algorithm complexity.

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