Dynamic Single Frequency Networks

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***Abstract*—Wireless asymmetric Internet access with a downlink peak bit rate of 10 to 30 Mb/s can be achieved by using the terrestrial digital video broadcasting system (DVB-T) as a supplemental downlink together with today's cellular systems. This paper is a study of dynamic radio resource management on a packet-by-packet basis for this broadband downlink. The dynamic single frequency networks (DSFN) scheme is evaluated. It exploits the macrodiversity capability of the OFDM modulation scheme. The transmitters are dynamically divided into groups of transmitters that send the same information at the same channel frequency simultaneously. The fairly shared spectrum efficiency (FSSE), in bits per second per Hertz per site, which is a combined measurement of maximum throughput and fairness, is evaluated for best-effort traffic. DSFN improves the FSSE by 100% to 370%, for a certain set of test cases, in comparison to the dynamic packet assignment (DPA) scheme, which combines packet scheduling with dynamic channel assignment (DCA).**

***Index Terms*—DPA, DSFN, DVB-T, fairness, FSSE, macrodiversity, OFDM, scheduling, SFN.**

I. Introduction

POPULAR Internet applications, such as WWW, Internet radio, and thin clients, are characterized by *asymmetric communication*, i.e., a much higher data rate to the user terminal than from it. Especially in wireless communication, the limited battery capacity makes high uplink data rates less interesting than high downlink rates. However, cellular communication systems for wide-area coverage (such as GSM and WCDMA) are not designed with the asymmetric communication in mind, since the uplink and downlink bands have equal capacity.

To increase the downlink capacity in the *general packet radio service* (GPRS) cellular system, a broadband *orthogonal frequency division multiplex* (OFDM) supplemental downlink was proposed in [1]. The proposal supports up to 10 Mb/s in microcellular environments, over 5-MHz wide channels. The OFDM modulation is chosen because of its ability to combat the frequency selective fading and *intersymbol interference* (ISI) due to multipath propagation, without the need of complex equalization.

A more evolutionary approach is to use broadband OFDM radio technology for wide-area coverage existing on the market today, instead of inventing a new air interface. The *digital audio broadcasting (DAB)* system Eureka 147 [2] has considerable coverage in Europe and Canada. The *terrestrial digital video broadcasting* system (*DVB-T*) [3] is rapidly expanding in

TABLE I
Technical Details for DAB and DVB-T

	DAB	DVB-T
Adopted:	1995	1997
Net bit rate R per frequency channel:	576 - 1152 kbit/s	4.98 - 31.67 Mbit/s
Channel separation B:	1.712 MHz	8 MHz
Link level spectrum efficiency R/B:	0.34 - 0.67 bit/s/Hz	0.62 - 4.0 bit/s/Hz
Freq. range of today's receivers:	174 -240 MHz, 1452 - 1492 MHz.	470 - 862 MHz
Maximum traveling speed:	About 200 - 600 km/h [4]	36 - 163 km/h (53 - 185 km/h) [a]
OFDM sub-carrier modulation:	DQPSK	QAM, 16QAM or 64QAM
Number of sub-carriers:	192, 384, 768 or 1536	1705 or 6817
Inner Forward Error Correction Coding (FEC):	Convolutional coding with code rates 1/4, 3/8 or 1/2	Convolutional coding with rates 1/2, 2/3, 3/4, 5/6 or 7/8.
Outer FEC:	None	RS(204,188,t=8)
Outer bit-interleaving (time interleaving):	Convolutional interl. of depth 384 ms	Convolutional interl. of depth 0.6 - 3.5 ms

[a] With modified receiver for FFT leakage equalization. [5]

Europe, Australia, and Asia. DAB can offer a net bit rate of 1.2 Mb/s. DVB-T offers about 24 Mb/s to stationary receivers with directional antennas and 12 Mb/s to mobile receivers with omnidirectional antennas. For technical details, see Table I.

Since infrastructure already exists, personal communications systems based on these technologies would require minimal initial infrastructure investments. It is expected that low-cost equipment will be available for these technologies in a few years and that radio spectrum will be free, especially if the analog TV transmissions are shut down in the end of this decade, in accordance with plans in some countries.

The EU ACTS project Multimedia Environment for MObiles (MEMO) [6] delivered a complete system specification for interactive services in the DAB system. This includes cellular Internet access, by using the DAB system as broadband downlink, and the GSM system as narrowband uplink. A combination of DVB-T and GSM was demonstrated in the *SABINA* pilot project (*System for Asymmetric Broadband INternet Access*) [7], [8], initiated by the Swedish national broadcasting company Teracom AB.

The *radio resource management* (RRM) in today's MEMO specifications for DAB is conventional. It is based on *fixed channel allocation* (FCA), with *frequency division multiple access* (FDMA), static handover criteria (i.e., static cell formations), and no power control [9], [10]. All DAB transmitters are always sending, irrespectively of if there is data to send or not. The RRM functions for the DVB-T case are not yet designed but are expected to be a further development of the RRM in MEMO.

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A major challenge in the design of cellular systems is to utilize the base station equipment and the limited frequency spectrum as efficiently as possible. A substantial improvement of the system level *spectrum efficiency* (maximum throughput in bits per second per Hertz per base station site) can be achieved by *dynamic RRM* techniques, such as *dynamic channel allocation (DCA), link adaptation* (i.e., change of modulation scheme and forward error correction coding) and *traffic adaptive handover* (also known as *cell breathing* or *load balancing*). A drawback with DCA is that it may require a large number of frequency channel units at every base station site. High spectrum efficiency in bits per second per Hertz per site, and thus dynamic schemes such as DCA, are desirable if at least one of the following conditions are satisfied: 1) the number of frequency channels is insufficient for making an FCA cell plan with sufficiently low cochannel interference level or 2) the base station site cost, alternatively the frequency cell planning cost, dominates over the cost for frequency channel units and additional system complexity due to DCA. The first criteria may be valid for a DVB-T-based cellular system in some regions today, and the second criteria is expected to be valid during future expansion of the system toward a micro-cellular structure.

The aim of this study is to propose and evaluate dynamic RRM schemes for nonrealtime packet mode communication over an OFDM-based downlink. The schemes are evaluated for the DVB-T case regarding spectrum efficiency, fairness, and computational complexity.

In today's digital cellular systems, interference fluctuations are handled by so-called *interference averaging*, e.g., frequency hopping or DS-CDMA. This is not possible in DVB-T, without a major change of existing hardware. However, that should not be considered as a problem. On the contrary, it was shown by Pottie [11] that *interference avoidance* by DCA and power reservation can perform a factor 2 to 3 better spectrum efficiency than interference averaging techniques.

Interference avoidance by resource reservation requires synchronized and centrally controlled base stations. Centralized resource reservation for packet mode communication is complex for a duplex system, but alluring in our case since the task only is to design RRM for the downlink. A centralized downlink system can gather information about the destinations of all data packets in the queues without a multiple access protocol.

The DVB-T system is designed to facilitate *single frequency networks (SFN)*, i.e., groups of transmitters sending the same information simultaneously over the same frequency channel, resulting in good coverage of a region and efficient frequency utilization for broadcasting services. The OFDM modulation scheme avoids interference due to this transmitter macro diversity, if the base station transmitters are sufficiently close.

In our previous work [12]–[14] we introduced the concept of *dynamic single frequency networks* (DSFN), where SFNs are utilized for personal communication services by adopting the SFN grouping to the receiver conditions. For an overview of DSFN, see Section II. For further details on the algorithms, see Section IV.

Packet mode cellular systems have a potential of high spectrum efficiency because of the ability to adapt the data rate for nonrealtime services to the interference level. In the packet data services for CDMA and TDMA-based cellular systems, this is handled by negotiating and allocating the radio resources before every burst of data packets. Thus the average interference is controlled on a *burst-by-burst* basis [15].

Since our task is to design RRM only for the downlink, it is possible to perform fast interference avoidance on a *packet-by-packet* basis, instead of on a burst-by-burst basis, without extensive wireless signaling. This means that we can even further utilize the bursty nature of packet mode communication by performing RRM for each data packet and each time slot individually and combine the RRM with *statistical multiplexing*, i.e., data packet scheduling.

The DSFN scheme performs a packet-by-packet RRM. A centralized scheduling algorithm changes the SFN grouping from time slot to time slot and assigns each data packet to an SFN and a time slot.

The system model and performance measures are defined in Section III. This includes a simplified model of best-effort traffic and a combined measurement of spectrum efficiency and fairness, called *fairly shared spectrum efficiency* (FSSE).

For reference, DSFN is compared with a fixed channel allocation (FCA) system with static handover, similar to today's MEMO standard, as well as with traffic adaptive handover, i.e., cell breathing; see Section V-A.

DSFN is also compared with the *dynamic packet assignment (DPA)* scheme [1], which was proposed by AT&T Labs for the OFDM downlink mentioned above. DPA performs DCA and scheduling of each data packet individually; see Section V-B.

Results and conclusions are presented in Sections VI and VII.

II. The Concept of Dynamic Single Frequency Networks (DSFN)

For pedagogical reasons, the basic principles of DSFN are presented before the system simulation model and performance measures are formally defined in Section III.

The base station transmitters are divided into SFNs, i.e., groups of transmitters that send the same information at the same channel frequency simultaneously. (The term SFN originates from the broadcasting world, where a network is a group of transmitters that send the same TV or radio program. In the cellular systems tradition, SFNs are referred to as a kind of *transmitter macrodiversity* or *simulcasting*.)

SFNs are facilitated by the OFDM modulation scheme, since OFDM avoids intersymbol interference (ISI) and coops with frequency-selective fading caused by this severe form of multipath propagation. SFNs are difficult to achieve with a conventional modulation scheme, since complex equalization would be required.

By the term DSFN we mean that the SFN grouping is changed from time slot to time slot and adopted to the receiver conditions. A large number of base station transmitters can be assigned to a receiver terminal in an exposed position, and thus cochannel interference can be avoided. The channel can be reused at shorter distance if the receiver is positioned nearby a transmitter.

A simple example: (See Fig. 1.) A system consists of two synchronized and centrally controlled base station transmitters,

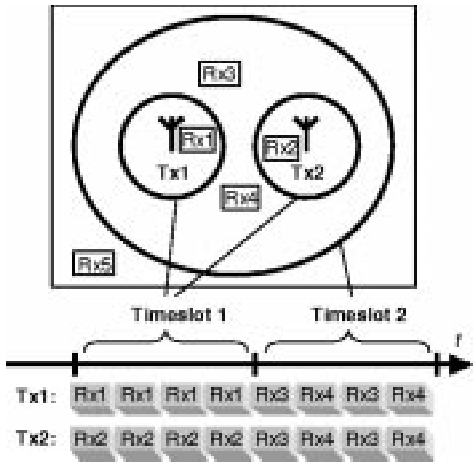

Fig. 1. A simple example of dynamic single frequency networks. Top: Coverage map. Below: Data packet schedule, stating the packet destinations.

Tx1 and Tx2, and five receiver terminals, Rx1 to Rx5, all assigned to the same frequency channel. The five receiver terminals are backlogged, i.e., there are data packets destined to them.

During the first time slot, Tx1 and Tx2 send different information. This can only be received within the two inner circles, since the cochannel interference level is too high outside the circles. The schedule shows that during time slot 1, Tx1 and Tx2 send data packets destined to receiver Rx1 and Rx2, respectively. During next time slot, both transmitters send the same information simultaneously, i.e., they are grouped to an SFN. The SFN covers the whole ellipse and can therefore send data packets destined to receiver Rx3 and Rx4. Receiver Rx5 cannot be covered and is in a state of *outage*.

The *spectrum efficiency* is R/B and $R/2B$ bit/s/Hz/transmitter site during timeslot 1 and 2 respectively, where R is the transmitter useful bit rate and B is the channel bandwidth. The spectrum efficiency averaged over the whole period is $3R/4B$ bit/s/Hz/site; see Section III-E.

The lowest average data rate that a receiver achieves during this example is $R/4$ (obtained by Rx3 and Rx4). If all four backlogged receivers that are not in outage would achieve this data rate, the spectrum efficiency would be $R/2B$ bit/s/Hz/site. This is what we call the FSSE; see Section III-F.

DSFN enables *soft handover*, meaning that when a receiver terminal moves from one base station transmitter toward another, both transmitters send to the receiver awhile instead of abruptly switching from the first to the second transmitter. Soft handover is robust toward sudden shadow fading of one of the transmitters.

DSFN is also a way of introducing time slots and DCA into DVB-T, without keying of the transmitter power. All transmitters continuously transmit at constant power, and the SFN grouping is changed from time slot to time slot. Thus, transmitter equipment existing today may be used.

DSFN simplifies the problem of packet-by-packet RRM substantially. Since all transmitters send continuously using constant power, all transmitters that are not assigned to a certain receiver can be considered as interferers. Hence, the interference level to a certain receiver can be analyzed without knowledge of the traffic assigned to other transmitters.

The OFDM scheme allows the receiver to measure the path loss from all neighboring transmitters simultaneously, by means of orthogonal transmitter identification codes, or by assigning different pilot subcarriers to different transmitters. Based on these measurements, a distributed algorithm executed in each receiver terminal identifies the *minimum SFN*, i.e., the set of transmitters that is required for sufficient SIR; see Section IV-A. The terminal reports the minimum SFN to a central *system controller*.

A centralized DSFN scheduling algorithm organizes the transmitters into SFNs separately for each time slot and frequency and assigns data packets to SFNs, time slots, and frequency channels; see Section IV-B. Note that SIR calculations are only performed locally in the terminals.

However, in this paper the scheduling scheme does not choose the frequency channel. Receiver-to-frequency channel assignment is assumed to be handled by a separate algorithm.

III. Models and Performance Measures

A. Wave Propagation Model

A system consists of a set $\mathbb{TX}$ of N_{Tx} centrally controlled and synchronized base station transmitters, sending information to the set $\mathbb{RX}$ of N_{Rx} receiver terminals, using the same frequency channel. The power from transmitter $i \in \mathbb{TX} = \{1, 2, \ldots, N_{Tx}\}$ at receiver $j \in \mathbb{RX} = \{1, 2, \ldots, N_{Rx}\}$ is modeled as

$$P_{i,j} = \frac{P_i F_{i,j} G_{i,j}}{d_{i,j}^{\alpha}} \tag{1}$$

where

- P_i transmitted power level from transmitter i;
- $d_{i,j}$ distance between the transmitter and receiver;
- $F_{i,j}$ depends on the antenna gains, antenna heights, and the carrier frequency;
- α propagation exponent;
- $G_{i,j}$ gain due to log-normal shadow fading.

The shadow fading gain $10\log_{10} G_{i,j}$ is normally distributed with expectation 0 dB and standard deviation σ_{S}.

In the simulation model, $\alpha = 4$ and $\sigma_{\mathrm{S}} = 8$ dB.

In this paper, no power control is considered. All transmitters are either sending at the same full power P_i, or blocked, i.e., $P_i = 0$.

Omnidirectional transmitter and receiver antennas are considered, implying that $F_{i,j}$ is constant and equal for all transmitters and receivers. In reality, some of the receiver antennas may be stationary directional UHF TV-antennas and sector antennas may be used in the transmitters to increase the spectrum efficiency. We have chosen this model since the system must be designed to handle the mobile case and since our task is to evaluate dynamic RRM algorithms and not cell planning strategies.

The N_{Tx} transmitters in a system are positioned on concentric hexagons, so that each transmitter has the same distance D to its six closest neighbors; see the example in Fig. 2. The centralized system is surrounded by the set $\mathbb{TX}^{\mathrm{C}}$ of external transmitter sites positioned at hexagons at distance D outside the outermost transmitter. Adjacent systems of transmitters use different frequency channels. A *large-scale handover algorithm*

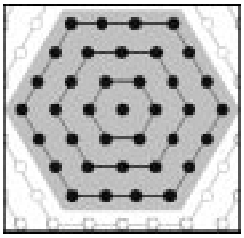

Fig. 2. Example of a system with $N_{Tx} = 37$ transmitters (filled circles) positioned at three concentric hexagons and surrounded by transmitters using other frequency channels (nonfilled). If $\sigma_S = 0$, the assigned receivers are uniformly distributed in the shaded service area.

assigns receiver j to the system if $P_{i,j} > P_{i',j}$ for $\forall i \in \mathbb{TX}$ and $\forall i' \in \mathbb{TX}^C$. The system *service area* is the set of geographical points that fulfills this large-scale handover criterion. The service area is shaded in the figure, for a case without shadow fading.

The N_{Rx} receivers that are assigned to the system are uniformly distributed within the service area of the system.

B. Single Frequency Networks and Link Quality Model

An SFN is a set of one or several transmitters sending the same information simultaneously over the same frequency channel. The signal-to-interference ratio (SIR) at receiver j, averaged over all OFDM subcarriers, is measured according to [16]

$$\Gamma_j = \frac{\sum_{i \in \mathbb{U}_j} P_{i,j} w_{i,j}}{\sum_{i \in \mathbb{I}_j} P_{i,j} + \sum_{i \in \mathbb{U}_j} P_{i,j}(1 - w_{i,j}) + I_{\mathrm{Ext},j}} \approx \frac{\sum_{i \in \mathbb{U}_j} P_{i,j}}{\sum_{i \in \mathbb{I}_j} P_{i,j} + I_{\mathrm{Ext}}} \tag{2}$$

where

$P_{i,j}$	power from transmitter i received in terminal j;
$\mathbb{U}_j \subseteq \mathbb{TX}$	set of transmitters in the SFN assigned to receiver j (the useful signals);
$\mathbb{I}_j = \mathbb{TX} \backslash \mathbb{U}_j$	set of transmitters assigned to other receivers (the cochannel interferers) in the centralized system;
$w_{i,j} \in [0, 1]$	weighting factor depending on the ISI and Doppler shift;
$I_{\mathrm{Ext},j}$	external interference power, including thermal noise as well as power from transmitters outside the centralized system.

$I_{\mathrm{Ext},j}$ is further discussed in Section III-D.

In our simulations, we neglect ISI and Doppler shift. This leads to the approximation in the last term of (2).

C. DVB-T System Parameters

Dynamic RRM require that *RRM parameters*, such as channel allocation, modulation, coding, and transmission power, can be changed at certain time instants. Therefore, a timeslot structure is introduced into the DVB-T system. RRM parameters can be changed between the time slots but not in the middle of a time slot. A time slot should consist of an integer number of OFDM symbols. The bit interleaving and error correction should not spread IP packets over several slots.

We assume that a DVB-T *frame*, consisting of 68 OFDM symbols, is used as the time slot entity, that the OFDM guard interval is 1/8 of the OFDM symbol duration, and that 1705 OFDM subcarriers are used. Thus, the timeslot is 17.136 ms.

The following DVB-T *transmission schemes*, i.e., combinations of modulation and error coding, are evaluated: QPSK modulation with code rate 1/2, QPSK 2/3, 16 QAM 1/2, 16 QAM 2/3, 64 QAM 1/2 and 64 QAM 5/6. These seven schemes are chosen because they give an integer number of MPEG transport stream packets per time slot, and have low bit error probability for a Rayleigh channel model.

These are referred to as scheme number $m = 1$ to 7. In this paper, no link adaptation is considered, i.e., m is the same for all transmitters.

Each transmission scheme is characterized by a *link bit rate* R_m (in bit/s) and an SIR bound γ_m. The RRM scheme strives at reserving resources such that $\Gamma_j > \gamma_m$ for all receivers. The SIR bound γ_m should include a margin for measurement data inaccuracy, e.g., due to terminal motion since last measurement. The margin should be based on the measured variance of the SIR, and chosen for a desired probability of TCP *automatic repeat request* (ARQ). A low ARQ probability is desirable, since the narrow-band back channel and the wired infrastructure may cause a long delay.

The SIR bounds γ_m used in the simulations correspond to bit error probability 10^{-11} for a Rayleigh fading channel and are obtained from [3, Annex A].

All IP packets have equal length of 1500 bytes (the maximum payload of Ethernet frames) in the simulations.

The seven transmission schemes can transfer 7, 10, 15, 20, 23, 30, and 38 IP packets per timeslot, respectively. Here the DVB-T convolutional interleaving is assumed to be modified to *block interleaving*, by a simple reordering of the information bytes before the interleaving and after the deinterleaving. This modification increases the useful data rate by between 0% and 11%, since less zero padding is required for avoiding that IP packets are spread over several timeslots.

D. External Interference Model and Outage Analysis

Receiver j is said to be in a state of *outage* if the RRM scheme is not able to assign resources to the receiver for sufficient SIR $\Gamma_j > \gamma_m$. In our simulation model, outage can only be caused by external interference (noise and interference from transmitters outside the centralized system), since the dynamic RRM scheme can avoid all internal interference to a vulnerable receiver. In the worst case, a DSFN scheme can assign all transmitters $i \in \mathbb{TX}$ to the receiver. Thus, the outage probability for a DSFN system is defined as

$$\chi_{\mathrm{DSFN}}(\gamma_m) \triangleq \Pr\left\{ \frac{\sum_i P_{i,j} w_{i,j}}{\sum_i P_{i,j}(1-w_{i,j}) + I_{\mathrm{Ext},j}} < \gamma_m \right\} \approx \Pr\left\{ \frac{\sum_i P_{i,j}}{I_{\mathrm{Ext}}} < \gamma_m \right\}. \tag{3}$$

A non-SFN DCA scheme such as DPA can avoid interference to a vulnerable receiver by only sending with one transmitter and block all other transmitters in the system. Thus, the outage probability for non-SFN systems has the following lower bound:

$$\chi_{\mathrm{NoSFN}}(\gamma_m) \geq \Pr\left\{ \frac{\max_i P_{i,j} w_{i,j}}{\sum_i P_{i,j}(1-w_{i,j}) + I_{\mathrm{Ext},j}} < \gamma_m \right\} \approx \Pr\left\{ \frac{\max_i P_{i,j}}{I_{\mathrm{Ext}}} < \gamma_m \right\}. \tag{4}$$

Consequently, $\chi_{\mathrm{NoSFN}}(\gamma_m) \geq \chi_{\mathrm{DSFN}}(\gamma_m)$, i.e., DSFN can improve the outage probability.

If the external interference is varying in time, I_{Ext} is defined as the maximum external interference that can occur, rather than the average external interference. The reason is that we do not average the interference level by spread spectrum technology, and thus the RRM scheme has to calculate SIR for the worst case.

A question may arise whether χ_{DSFN} and χ_{NoDSFN} are comparable. Specifically: Would the external interference be the same if the external transmitters were continuously sending, for example a DSFN system, as if they were belonging to a noncontinuously transmitting system such as a DPA? The answer is yes because the RRM scheme must calculate with the worst case interference level. The worst case scenario corresponds to that all external transmitters are sending at full power, in the noncontinuous as well as the continuous case.

If the external interference were neglected in the simulations, the system behavior would deviate considerably from a real world system. The outage probability would be zero, since RRM schemes such as DSFN and DPA can avoid all internal interference to a weak receiver, corresponding to infinite SIR. The spectrum efficiency could be very high since it is possible to use a high code rate and a large number of modulation symbols without interference problems.

The external interference level is modeled as homogeneous, i.e., the same $I_{\mathrm{Ext},j} = I_{\mathrm{Ext}}$ for all receivers j. An argument for this simplification is that the system is very sensitive to external interference, since external interference is not avoided or spread by the RRM schemes. For a reasonable outage level the external interferers must be positioned at far distance, meaning that the distance is similar to each receiver in the system.

Only transmission schemes m that provide outage $\chi < 5\%$ are considered.

The outage due to external interference is a cell planning issue and cannot be controlled by the dynamic RRM algorithms. Because of this, we assume that the external interference has a level corresponding to an outage $\chi_{\mathrm{DSFN}}(\gamma_{\mathrm{Ref}})$ of 5%, for a certain *reference SIR boun*d $\gamma_{\mathrm{Ref}} = \gamma_6 = 19.3$ dB.

E. A Best-Effort Traffic Model for Spectrum Efficiency Analysis

Only nonrealtime *best-effort traffic* is considered, i.e., delay insensitive communication without differentiated priorities or QoS guarantees.

The following model aims at simplifying the evaluation of maximum throughput and fairness for best-effort traffic.

An *active receiver* is a terminal with at least one data packet waiting in the system queues (in the literature sometimes called a *backlogged* terminal), and that is *not* in outage.

$\mathbb{RX}$ is the set of N_{Rx} active receivers.

There is a density of ω backlogged receiver terminals per transmitter site in the system. Note that this figure includes receivers that are in outage.

During a period of terminal activity, the system transfers a *data burst* to the terminal. The *average data rate* (in bits per second) that terminal j acquires during the data burst is denoted r_j. This is the maximum *user throughput* that the system can deliver. A terminal that does not require all of this available bit rate in the long run will rapidly alternate between active and passive state.

The system level *spectrum efficiency* in bits per second per Hertz per transmitter site of the system is a normalized measure of the average user throughput and is defined as

$$\eta(\omega) \triangleq \frac{1}{N_{\mathrm{Tx}}B} \mathrm{E}\left[\sum_{j \in \mathbb{RX}} r_j\right] \approx \frac{\omega(1-\chi)}{B} \mathrm{E}_{j \in \mathbb{RX}}[r_j] \tag{5}$$

where B is the available radio spectrum bandwidth.

In the simulations, a *snapshot* or *steady-state* model is used. A constant set of stationary active terminals is used in each simulation, i.e., no data bursts are initiated or finished during a simulation. All users are constantly in a maximum throughput situation.

When several RRM schemes are compared, a more efficient scheme would result in higher user data rates but unchanged density ω in back-logged terminals/transmitter. There are several reasonable interpretations of this assumption: 1) The user behavior is affected by the increased performance of the RRM scheme, such that the user communicates the same amount of time, but transfers more data; 2) The market is affected by the increased performance, so the amount of subscribers is increased. Each user transfers the same amount of data, but at shorter time; and 3) The increased performance makes it possible for the service provider to position the transmitters less dense. The number of subscribers is unaffected. Each user transfers the same amount of data, but at shorter time.

F. Fairness Analysis

We propose a combined performance measure of fairness and spectrum efficiency, which we call the *fairly shared spectrum efficiency* (FSSE) in bits per second per Hertz per transmitter site, and define

$$F(\omega) \triangleq \frac{1}{N_{\mathrm{Tx}} B} \mathrm{E}\left[N_{\mathrm{Rx}} \min_{j \in \mathbb{RX}} r_j \right] \approx \frac{\omega(1-\chi)}{B} \mathrm{E}\left[\min_{j \in \mathbb{RX}} r_j \right]. \tag{6}$$

This can be described as a normalized measure of the minimum throughput that a terminal achieves or a measure of the part of the total system capacity that is equally shared among all active terminals.

Dynamic RRM in combination with best-effort traffic may cause absurd unfairness, if it is designed only with the objective to maximize the spectrum efficiency η (and by that the average user throughput and total system throughput). If several active terminals were contending about the same transmitter, the spectrum efficiency would be maximized if the least "expensive" terminal (e.g., the terminal at shortest distance from the transmitter) were allowed to use the whole resource without sharing it with the others. Some terminals would suffer from starvation, corresponding to $F(\omega) = 0$. The network service provider may lose income from making customers unsatisfied due to this unstable service quality.

On the other hand, equal resource sharing such that $\eta(\omega) = F(\omega)$ is a waste of resources. We do not want to prevent a terminal from using a free time slot because it has already achieved higher data rate than other terminals, although it is impossible to assign the slot to any other terminal.

We strive at *max–min fairness* [17], which is a widely accepted compromise between the two above extreme strategies. Max–min fairness implies that the first priority is to maximize the lowest average data rate r_j that an active terminal achieves, the second priority is to maximize the second lowest data rate, etc. The data rates are max–min fair if and only if no data rate r_j can be increased without forcing a decrease in another rate of equal or lower value.

In a max–min fair system, FSSE is maximal. This is the motivation for the FSSE performance measure.

IV. The DSFN Algorithms

The basic principles of DSFN were introduced in Section II. Details on the algorithms are presented in this section.

A. Distributed Algorithm for Identification of the Minimum SFN

The *minimum SFN* $\mathbb{M}_{j,m}$ of receiver j and transmission scheme m is the minimum set $\mathbb{M}_{j,m} \subseteq \mathbb{TX}$ of transmitters assigned to the receiver, such that the required SIR γ_m is achieved.

We propose the following distributed algorithm for the identification of the minimum SFN $\mathbb{M}_{j,m}$. Start with an empty set $\mathbb{M}_{j,m}$. Extend $\mathbb{M}_{j,m}$ by the nonused transmitter that gives highest received power iteratively, until the required SIR bound γ_m is achieved. If the terminal requires a bigger SFN than the number of centrally controlled transmitters, the algorithm indicates that the terminal is in a state of *outage* by setting $\mathbb{M}_{j,m}$ to the empty set.

The algorithm pseudocode follows:

$\mathbb{M}_{j,m} := \varnothing$
do
  if $\mathbb{M}_{j,m} \neq \mathbb{TX}$
    $\mathrm{M}_{j,m} := \mathrm{M}_{j,m} \cup \arg\max_{i \in TX \setminus M_{j,m}} P_{i,j}$
  else {
    $\mathbb{M}_{j,m} := \varnothing$
    break
  }
while $\sum_{j \in \mathrm{M}_{j,m}} P_{i,j} \Big/ \left(P_{\mathrm{Tot},j} - \sum_{j \in \mathrm{M}_{j,m}} P_{i,j} \right) < \gamma_m$.

□

The function $\arg\max_i x_i$, here applied on a vector (x_i), is defined as the index i of the largest element x_i, or the index to the first of several equal elements with the largest value.

$P_{\mathrm{Tot},j}$ is the total received power in transmitter j, i.e., the sum of the nominator and denominator of (2).

If link adaptation should be supported, the algorithm is repeated for each transmission scheme m.

It can be proven that the algorithm minimizes the SFN size for a required SIR. Note that this is not always the same thing as maximizing the spectrum efficiency or FSSE. Those performance measures are expected to be improved if the algorithm is modified to avoid using transmitters with high load, i.e., to perform traffic adaptive load balancing.

B. Centralized DSFN Scheduling Algorithm

Each receiver j reports the minimum SFN $\mathbb{M}_{j,m}$ to the central system controller. A centralized scheduling scheme can assign data packets destined to terminal j to time slot t and SFN $\mathbb{U}_j$.

Reference [18] shows that the *max–min fairness* objective described in Section III-F can be achieved by employing *fair queuing*, for example based on a *weighted round robin* algorithm. This comprises that the central system controller has a separate first-come first-served data packet queue for each receiver. L_j denotes the number of equally sized packets in the queue destined to receiver j.

The scheduling algorithm is performed once per time slot.

The *input parameters* to the scheduling algorithm are: the set of centrally controlled transmitters $\mathbb{TX}$; the transmission scheme m; the minimum SFNs $\mathbb{M}_{j,m}$; the number of equally sized packets L_j in the queue destined to receiver j (the queue length); and the maximum number of data packets per time slot R'_m.

The *output parameters* of the scheduling algorithm are: number of scheduled data packets $\mathrm{NP2Rx}_j$ to each receiver during the time slot; the *SFN to receiver assignment vector*

$$\mathrm{SFN2Rx}_j \triangleq \begin{cases} n, & \text{if receiver } j \text{ is assigned to SFN number } n \\ 0, & \text{if receiver } j \text{ is not assigned to this timeslot} \end{cases} \tag{7}$$

and the *SFN to transmitter assignment vector*

$$\mathrm{SFN2Tx}_i \triangleq \begin{cases} n, & \text{if transmitter } i \text{ is assigned to SFN } n \\ 0, & \text{if transmitter } i \text{ is not used.} \end{cases} \quad (8)$$

The set $\mathbb{RX}$ of active receivers can be identified as

$$\mathbb{RX} \triangleq \{j\colon L_j > 0 \wedge \mathbb{M}_{j,m} \neq \varnothing\}. \quad (9)$$

A modified fair queuing algorithm is presented here. It gives scheduling priority to the terminals that have acquired lowest data rate since they became active.

Introductory simulations show that it is beneficial to schedule the terminals with biggest minimum SFN first. An intuitive explanation is that it is easier to pack a knapsack efficiently if we start with the big objects and put small objects in spaces in between. Thus, the SFN size also affects the scheduling priority in the algorithm.

The scheduling may not be efficient if we always use the minimum SFN to each receiver, such that $\mathbb{U}_j = \mathbb{M}_{j,m}$. For example, it may be beneficial to send packets to two terminals j_1 and j_2 with similar minimum SFNs during the same time slot. Then the scheduling scheme should combine the two minimum SFNs, such that $\mathbb{U}_{j_1} = \mathbb{U}_{j_2} = \mathbb{M}_{j_1,m} \cup \mathbb{M}_{j_2,m}$. However, introductory simulations indicate that for the steady-state traffic model with unlimited number of packets to each terminal, $\mathbb{U}_j = \mathbb{M}_{j,m}$ is the most efficient solution. A combination of SFNs is only advantageous when the queue to a terminal becomes empty before the time slot ends and there is room for more packets in the schedule to a similar SFN during the same time slot. Because of this, the scheduling algorithm has two phases. Phase 1 only assigns terminals with disjoint or equal minimum SFNs. If a queue becomes empty before phase 1 has come to an end, phase 2 will be carried out, which tries to combine several minimum SFNs. Note that phase 2 cannot be evaluated for our traffic model.

The algorithm counts the number of packets C_j that each terminal j has sent. The algorithm iteratively tries to schedule a packet to the receiver that has first minimum counter value C_j and secondly the biggest minimum SFN. When a terminal j enters the active state its counter is set to

$$C_j := \max\left(C_j,\ \min_{j' \in \mathbb{RX} \setminus j} C_{j'}\right). \quad (10)$$

Thus a packet from a new data burst achieves highest priority. A terminal that recently left the active state is prevented from getting a more advantageous place.

To avoid counter overflow, all counters may be adjusted, e.g., after each time slot, according to

$$C_j := C_j - \min_{j' \in \mathbb{RX}} C_{j'}, \qquad \forall j. \quad (11)$$

Pseudocode for the DSFN scheduling algorithm follows:

$\mathrm{SFN2Tx}_i := 0, \forall i \in \mathbb{TX}$
$\mathrm{NP2Rx}_i := 0, \forall j \in \mathbb{RX}$
$\mathrm{NP2SFN}_n := 0, \forall n \in \mathbb{TX}$
$N_{\mathrm{SFN}} := 0$
$\alpha := \dfrac{1}{1 + \max_{j \in \mathbb{RX}} |\mathbb{M}_{j,m}| - \min_{j \in \mathbb{RX}} |\mathbb{M}_{j,m}|}$
$phase2\_flag := \mathrm{false}$

```
// Phase 1:
A := RX
while A ≠ ∅ {
  j := argmin_{j'∈A} (C_{j'} − α|M_{j',m}|)
  if NP2SFN_{SFN2Tx_i} = R'_m for any i ∈ M_{j,m}
    A := A\j
  else {
    if SFN2Tx_{j'} = 0, ∀ j' ∈ M_{j,m} {
      N_SFN := N_SFN + 1
      n := N_SFN
      SFN2Tx_i := n, ∀ i ∈ M_{j,m}
    }
    else
      n := SFN2Tx_i for one of the elements
  i ∈ M_{j,m}
    if ((n ≠ 0)
        and (SFN2Tx_i = n for any i ∈ M_{j,m})
      and (SFN2Tx_i ≠ n
      for any i ∈ TX\M_{j,m})) {
      SFN2Rx_j := n
      NP2Rx_i := NP2Rx_i + 1
      NP2SFN_n := NP2SFN_n + 1
      C_j := C_j + 1
      L_j := L_j − 1
      if L_j = 0 {
        A := A\j
        phase2_flag := true}
    }
    else A := A\j
  }
}

// Phase 2:
if phase2_flag {
  A := RX
  while A ≠ ∅ {
    j := argmin_{j'∈A} (C_{j'} − α|M_{j',m}|)
    if ((L_j = 0)
      or (NP2SFN_{SFN2Tx_i} = R'_m
      for any i ∈ M_{j,m}))
      A := A\j
    else {
      N := ⋃_{i∈M_{j,m}} SFN2Tx_i
      if N\0 consists of one element {
        n := N\0
        SFN2Tx_i := n, ∀ i ∈ M_{j,m}
        SFN2Rx_j := n SFN2Rx_j := n
        NP2Rx_i := NP2Rx_i + 1
        NP2SFN_n := NP2SFN_n + 1
        C_j := C_j + 1
        L_j := L_j − 1
      }
    }
  }
}
```

□

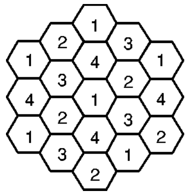

Fig. 3. FCA and DPA with $K = 4$.

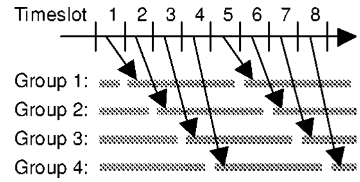

Fig. 4. The staggered DPA algorithm, for $K = 4$. During timeslot 1, transmitters in group 1 can reserve timeslots 2 to 5 after checking the interference. During slot 2, group 2 can schedule slots 3 to 6, etc.

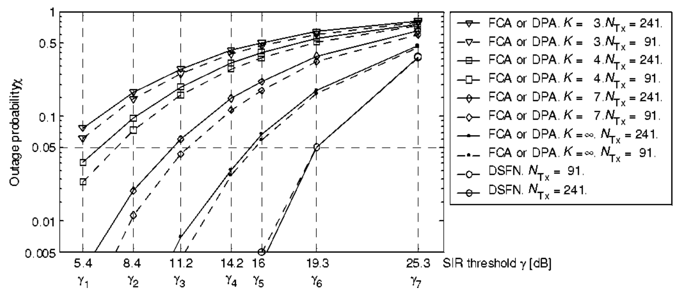

Fig. 5. Outage probability χ as a function of the SIR bound γ_m of the 7 modes m. Solid and dashed curves represent large and small system sizes. FCA and DPA have the same outage for the same K. The lower bound for DPA and FCA is denoted with $K = \infty$.

V. Reference Schemes

A. Fixed Channel Allocation (FCA)

For reference, conventional cellular FCA is evaluated. To make the results comparable with the DSFN evaluation, only one frequency channel is considered. Each base station transmitter is assigned to one of K TDMA channels and transmits during the corresponding timeslot independently of if there is something to send or not. *Reuse factors* of $K = 3, 4, 7, 9$, and 12 are considered; see the example of transmitter to channel assignment in Fig. 3.

Two handover schemes are evaluated: 1) *SIR-based static HO*, which assigns each receiver to the transmitter that provides maximum SIR. This is similar to today's MEMO system; and 2) *Traffic adaptive HO* (also known as cell breathing or load balancing): If a new active receiver belongs to overlapping cells (i.e., the SIR $> \gamma_m$ for several transmitters), it is assigned to the cell i with lowest number of assigned active receivers A_i. For each iteration of the algorithm, every active receiver that belongs to overlapping cells is checked. The receiver is reassigned to the cell i with the lowest number of active receivers A_i if the A_i values of the overlapping cells differ by two or more. If A_i differ by one, then the receiver is reassigned to the other cell with a certain probability, for example 50%, in view to make room for HO from more loaded cells near the first cell. For the steady-state traffic model, the variance $\mathrm{var}(A_i)$ is either unchanged or decreased after each iteration of this algorithm and converges within a few iterations.

B. Dynamic Packet Assignment (DPA)

The AT&T Labs network assisted DPA [1] is a combination of DCA and statistical multiplexing, i.e., data packet scheduling. The algorithm assigns transmitters and data packets to timeslots.

A traditional HO scheme assigns each terminal to a transmitter. We use the SIR-based static HO scheme above. The base station transmitters belong to K groups, where the transmitters in one group are nonadjacent. An example of a group division is given in Fig. 3.

DPA is based on a staggered scheduling algorithm, with the purpose to facilitate a distributed execution of the scheduling in each base station, without contention among adjacent transmitters; see Fig. 4. During timeslot n, the base stations in group $n \bmod K$ calculate if it will be possible for them to transmit during timeslot $n+1$, $n+2$, $\ldots$, $n+K$ without causing outage of already scheduled terminals. They also calculate if the active receivers that are assigned to them can capture the transmitter signal during each of these K slots. Finally, the algorithm assigns transmitters and data packets to the timeslots, and the base stations inform each other about their scheduling decisions by means of a fast backbone network.

A drawback is that DPA requires an SIR bound margin for interference among transmitters in the same group.

In the original proposal $K = 4$, but we evaluate other values up to $K = 12$. Thus the SIR bound margin can be reduced, and less robust transmission schemes can be used. In the original DPA, only one receiver is assigned to each timeslot and transmitter. Since our system can transfer many IP packets per timeslot, we modify DPA to allow several different receivers to share the same slot, to restrict the packet delay. *Fair scheduling* of each transmitter queue is added to the algorithm, such that scheduling priority is given to the receiver terminal j that has achieved lowest data rate r_j.

VI. Simulation Results

A. Outage

Fig. 5 illustrates that DSFN has considerably better outage probability χ than the other schemes, and that DSFN is allowed to use transmission schemes $m = 1$ to 6 for $\chi < 5\%$.

DPA with K groups has the same outage as FCA with reuse factor K. The lower bound given by (4) is denoted $K = \infty$.

DPA and FCA can use scheme $m = 1$ to 4 for $\chi < 5\%$.

The figure shows that our I_{Ext} model makes χ quite insensitive to the number of transmitters N_{Tx} in the system.

B. Spectrum Efficiency and Fairness

Fig. 6 shows the spectrum efficiency (left) and FSSE (right) for a system size of $N_{\mathrm{Tx}} = 91$, when transmission scheme m and (in the DPA and FCA cases) the factor K are chosen for maximum spectrum efficiency. Fig. 7 shows the corresponding results for $N_{\mathrm{Tx}} = 241$.

The highest spectrum efficiency $\eta(\omega)$ that is achieved for the simulated cases is 0.72 bit/s/Hz/site by DSFN.

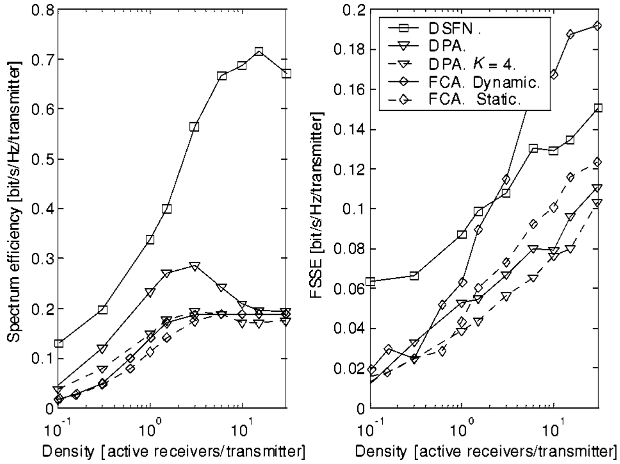

Fig. 6. Maximum throughput policy for selection of transmission scheme m and reuse factor K. $N_{\mathrm{Tx}} = 91$.

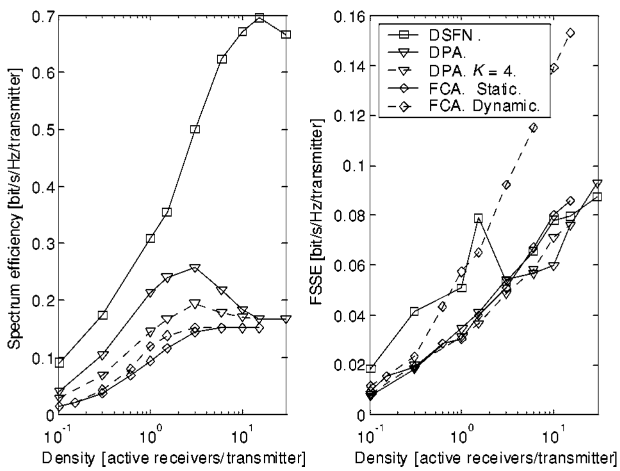

Fig. 7. Maximum throughput policy for selection of transmission scheme m and reuse factor K. $N_{\mathrm{Tx}} = 241$.

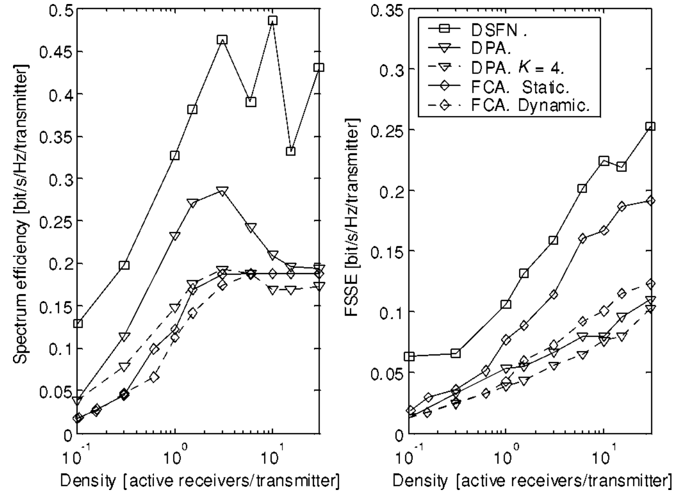

Fig. 8. Max–min fairness policy for selection of transmission scheme m and reuse factor K. $N_{\mathrm{Tx}} = 91$.

TABLE II
DSFN IMPROVEMENT RELATIVE TO THE OTHER SCHEMES

		FCA with static HO	FCA with adaptive HO	DPA
M	$\eta(\omega)$	156 to 651%	134 to 596%	45 to 313%
M	$F(\omega)$	-48 to +309%	-28 to +228%	-6 to +453%
F	$\eta(\omega)$	76 to 651%	76 to 596%	10 to 233%
F	$F(\omega)$	21 to 309%	17 to 228%	101 to 374%

TABLE III
COMPUTATIONAL COMPLEXITY

		$N_{Tx} = 19$		$N_{Tx} = 127$		$N_{Tx} = 241$
		$\omega = 1$	$\omega = 10$	$\omega = 1$	$\omega = 10$	$\omega = 1$
CPU time per packet (ms)	DPA	0.28	0.31	0.17	0.29	0.17
	DSFN	0.24	0.49	0.35	1.38	0.45
	%	-10%	+60%	+100%	+370%	+160%
Clock freq. (GHz)	DPA	0.3	0.4	0.9	1.5	2.2
	DSFN	0.5	2.1	3.4	31.8	7.5

The spectrum efficiency is not very sensitive to the system size N_{Tx}, but the FSSE is affected.

The jittery curves can be explained by the fact that m and K depend on the densities.

From the FSSE plots we can draw the conclusion that this *maximum spectrum efficiency policy* for choosing scheme m and K may result in impaired FSSE of DPA and DSFN in comparison to FCA. Thus, some users may experience worse performance if dynamic schemes are introduced.

In Fig. 8, a *max–min fairness policy* is adopted, such that m and K are chosen for maximum $F(\omega)$. The plot indicates that this policy lets every user experience that DSFN gives highest performance. $F(\omega)$ of up to of 0.25 bit/s/Hz/site is achieved by DSFN. This policy results in a more robust transmission scheme m.

Table II shows the performance improvement span of DSFN relative to the other schemes, for the maximum throughput (M) and max–min fairness (F) policies, evaluated for $N_{\mathrm{Tx}} = 91$ and 241, and ω in the range from 0.1 to 30. The highest improvement in percentage points is achieved at the lowest density ($\omega = 0.1$).

C. Computational Complexity

The computation complexity of the DSFN and DPA scheduling algorithms has been evaluated, in view of examining if software implementation of packet-by-packet RRM is feasible on a central computer that controls one system of transmitters and one frequency channel.

The execution time of the compiled Matlab 6.0 code has been timed on a 400 MHz Pentium II processor. Note that considerably better performance is expected for handwritten C code. Also note that only the time for producing the schedule is calculated. Time for actual data transfer, for managing measurement data, etc., is not considered.

Table III shows the average computation time per IP packet and the time percentage increase of DSFN relative to DPA. It also shows the required processor clock frequency in gigaHertz for real time execution, if we assume that the computation time is inversely proportional to the clock frequency.

Observe that DPA is more sensitive to the density than DSFN. DPA should be modified to only consider a subset of the receivers if the density is high.

VII. Discussion

Dynamic RRM schemes such as DSFN and DPA should be utilized with care for best-effort traffic. If fairness is not considered, large average user throughput in bits per second may be achieved, but several users may suffer from "starvation" and might be locked out from the system. Although fair scheduling is included in our algorithms and starvation is avoided, some users may achieve worse performance with the dynamic schemes than with FCA with static handover, if system parameters such as modulation and coding scheme are chosen for maximum spectrum efficiency.

If the system parameters are chosen according to a max–min fairness policy, DSFN not only achieves considerably higher average throughput than FCA and DPA, but also higher throughput to every user.

A major contribution of this study is the analysis of the FSSE. This measure is maximized in a max–min fair system.

The computation time analysis indicates that software implementation of the packet-by-packet RRM schemes on a central computer is feasible. However, the DSFN scheme should be modified to handle large user terminal densities.

An important conclusion of this study is that the advantageous effects of DSFN on the performance dominates over the disadvantageous effects.

An advantage with DSFN is that it can assign a large number of transmitters to a "weak" receiver terminal. This can be described as "extended power control," which allows higher power than the power from a single transmitter. Thus, less robust and more efficient modulation and error coding can be used for the same outage probability as a non-DSFN system. Especially when there is a low user density, we can afford to use less robust modulation and coding, since DSFN can assign large SFNs to most users without interference.

However, that cannot explain all of the gain due to DSFN, since DSFN also gives higher performance than the other schemes when it is restricted to the most robust modulation and coding scheme. An explanation is that DSFN facilitates the design of an efficient combined scheduling and RRM.

The reduced DPA performance for high densities is caused by the fact that DPA is modified to allow several receiver terminals to share a timeslot.

A disadvantageous effect of DSFN is shortly described in the following: Consider a DSFN system with SFNs of equal size K in a regular pattern. Also consider an equally large FCA system with reuse factor K. The spectrum efficiency is the same if the same modulation and coding is used, but SIR is lower in the DSFN case. The received useful power is less than K times higher in DSFN than in FCA, due to longer distances to the transmitters. The received interference power is more than K times stronger in DSFN than in FCA since there is a shorter distance to the nearest interferer in DSFN.

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