Transmit Power Performance Improvement Using Cooperative Multiuser OFDMA Systems

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Abstract— **In this paper we look into the performance of cooperative OFDMA systems under realistic conditions. We propose a transceiver structure to reduce the interference between transmitting subcarriers and receiving subcarriers. Its performance in terms of Signal to Interference and Noise Ratio (SINR) is evaluated by both analysis and simulation and is incorporated into a recently proposed cooperation strategy for Orthogonal Frequency-Division Multiple-Access (OFDMA) systems to examine its performance under the realistic structure.**
It is shown that although the cooperation strategy suffers from performance degradation due to the residual interference between the transmitting and receiving subcarriers, it still outperforms the conventional cooperation schemes.The effects of multi-path fading are critical in cellular, Ad-hoc and sensor networks because the physical deployment of communicating nodes makes them susceptible to interference. It is thus imperative to adopt a mechanism to combat fading in such networks. The broadcast nature of the wireless medium makes it easy for communicating nodes to hear each other, thus they can share their physical resources specifically their single antennas, thereby creating a virtual multiple-antenna array.

Keywords- **Signal to Interference and Noise Ratio (SINR), Orthogonal Frequency-Division Multiple-Access (OFDMA)**

I. INTRODUCTION

The fundamental idea of cooperative communications in wireless networks originates from the design of multiple antenna systems. In multiple antenna systems, communicating terminals are equipped with multiple antennas to mitigate the effects of multipath fading and optimize the communication rate in the network. Wireless communication nodes like cellular phones and sensor nodes have size restrictions, power supply limitations, and are only able to accommodate a limited level of complexity. It is thus unfeasible to equip them with multiple antennas. The effects of multi-path fading are critical in cellular, Ad-hoc and sensor networks because the physical deployment of communicating nodes makes them susceptible to interference. It is thus imperative to adopt a mechanism to combat fading in such networks. The broadcast nature of the wireless medium makes it easy for communicating nodes to hear each other, thus they can share their physical resources specifically their single antennas, thereby creating a virtual multiple-antenna array. Cooperative communications is a concept that enables several single-antenna nodes to cooperate and form a distributed multiple-antenna system which combats multipath fading. In recent years, cooperative communications has received extensive world-wide attention and significant advances in both research and applications have been achieved. Orthogonal Frequency Division Multiplexing (OFDM), essentially identical to Coded Orthogonal Frequency Division Multiplexing (COFDM) and Discrete Multi Tone (DMT) modulation, is a Frequency Division Multiplexing (FDM) scheme used as a digital Multi Carrier Modulation (MCM) method. Orthogonal Frequency Division Multiplexing (OFDM) is a method of Digital Modulation in which a signal is split into several narrowband channels at different frequencies. The OFDM technology was first conceived in the 1960s and 1970s during the research into minimizing interference among the channels near each other in frequency.

The main idea behind the OFDM is that since low-rate modulations are less sensitive to multipath, the better way is to send a number of low rate streams in parallel than sending one high rate waveform. This can be exactly done in OFDM. The OFDM divides the frequency spectrum into sub-bands small enough so that the channel effects are constant (flat) over a given sub-band. Then a classical IQ modulation (BPSK, QPSK, M-QAM, etc) is sent over the sub-band. If designed correctly, all the fast changing effects of the channel disappear as they are now occurring during the transmission of a single symbol and are thus treated as flat fading at the received.

The difficulty of implementing the schemes lies in the requirement that nodes are able to transmit and receive at the same time using adjacent subcarriers, which is referred to as “subcarrier-based duplexing” in the remainder of this paper. Therefore it is important to investigate the feasibility of subcarrier-based duplexing to determine the tradeoffs that may occur in the cooperative schemes . A small body of research has already proposed various sub carrier based duplexing schemes . Although subcarrier based duplexing appears possible in ideal OFDM systems, the orthogonality between different subcarriers is lost in realistic communication systems due to the non-ideal characteristics of different subsystems (e.g., frequency offset of local oscillator, nonlinearity of power amplifier, etc.) and these effects need to be addressed to understand how the transmitting subcarriers will interfere with neighboring receiving subcarriers. The idea of subcarrier-based duplexing was first proposed for digital subscriber line systems. It was later proposed to interleave subcarriers of an OFDM system for transmission and reception. Additionally, introduced a scheme where the available subcarriers are allocated in blocks, with an analog filter bank employed to isolate the desired receive signal from the transmitting signal and to reduce the signal dynamic range. further considers the issue of time synchronization, including the alignment of OFDM symbols and re-establishment of the FFT window. The utilized baseband echo cancellation to suppress the interference between transmitting and receiving subcarriers. Moreover, the 802.16 standard also incorporates relay stations that can transmit and receive simultaneously on different subcarriers .

In this paper our interest lies in the performance of cooperative OFDMA systems under subcarrier-based duplexing and in particular the tradeoffs and limitations in realistic configurations. To perform this we make use of a transceiver structure that utilizes baseband echo cancellation to suppress the interference between the transmitting and receiving sub carriers. The performance of this transceiver is verified by analysis and computer simulation, and it is shown that it is possible to achieve subcarrier-based duplexing in short-range low-transmit-power communication systems (e.g., 802.11a/g systems) with careful design. This scheme is then incorporated into the cooperation strategy of to investigate its performance under realistic conditions. It is revealed that although the performance of the cooperative network is degraded due to the residual interference imposed on the receiving subcarriers by the transmitting subcarriers, it still performs better compared with conventional cooperation schemes.

II. DSPs versus Microprocessors

DSPs differ from microprocessors in a number of ways. Microprocessors are typically built for a range of general purpose functions, and normally run large blocks of software, such as operating systems like Windows or UNIX. Although today's microprocessors, including the popular and well-known Pentium family, are extremely fast-as fast or faster than some DSPs--they are still not often called upon to perform real-time computation or signal processing. Usually, their bulk processing power is directed more at handling many tasks at once, and controlling huge amounts of memory and data, and controlling a wide variety of computer peripherals (disk drive, modem, video display, etc). However, microprocessors such as Pentiums are notorious for their size, cost, and power consumption to achieve their muscular performance, whereas DSPs are more dedicated, racing through a smaller range of functions at lightning speed, yet less costly and requiring much less space (size) and power consumption to achieve their purpose. DSPs are often used in "embedded systems", where they are accompanied by all necessary software (stored in onchip ROM or offchip EEPROM), built deep into a piece of equipment, and dedicated to a group of related tasks. In computer systems, DSPs may be employed as attached processors, assisting a general purpose host microprocessor.

Emerging Application: DSP-based Voice-over-Internet Protocol

Voice-over-Internet Protocol (VoIP) is one example that will benefit from TI's DSPs, particularly the new 'C54xx devices. Shifting voice traffic from traditional telephone networks to well-managed IP-based data networks is a growing trend that provides advantages such as system efficiency and lower calling costs. DSPs are at the core of the voice and data networks infrastructure, and TI, the world leader in DSP’s, is also the DSP leader in the VoIP market. A DSP-based VoIP system or gateway makes a shift to data networks possible, serving as the bridge between the Public Switched Telephone Network (PSTN) and the packet network. VoIP gateways allow users to speak on regular phones or send information over regular fax machines as they bypass PSTN toll charges with no perceivable loss of quality. DSP advancements in processing horsepower, smaller footprint and reductions in power dissipation have expanded the number of channels carried on VoIP gateways and embedded in network backbones. These advancements are transforming a technology, once used primarily to obtain free phone calls through a PC and the public Internet.

Sampling Theorem

According to sampling theorem, an analog signal can be exactly reconstructed from its samples if the sampling rate is at least twice the highest frequency component present in the signal. This means, if the signal contains the highest frequency component of 1 KHz, the sampling rate must be at least 2 Kilo Samples/ Second. This sampling rate is also known as Nyquist rate. What if sampling theorem is not satisfied..? Violation of sampling theorem means, sampling the signal at a sample rate less than Nyquist rate. This leads to unacceptable distortion in the signal. This distortion is known as aliasing. Due to aliasing, the components present in the input signal whose frequency is higher Nyquist frequency will be sampled as a signal of lower frequency. And this aliased samples in turn corrupts the original signal which is lower than Nyquist frequency. Due to the presence of aliasing, a signal that contains frequency components higher than Nyquist frequency cannot be reconstructed from its samples. The effect of aliasing is shown in the figure below.

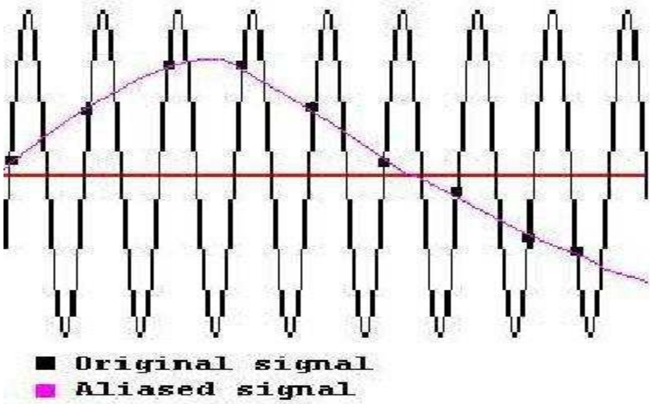

Fig 1 Effect Of Aliasing

The signal shown in black color is the original signal. While the signal shown in violate color is the aliased signal because of improper sampling. From the figure it is obvious that the aliased signal will be present in the sampled data as a lower frequency signal. And this will affect the original content at that frequency.

It should be noted that the information in between two consecutive samples is lost forever. But still the entire signal can be reconstructed from the samples as long as sampling theorem is satisfied.

Sampling and Quantization

Sampling can be viewed theoretically as multiplying the original signal with a train of pulses with unit amplitude. This leaves the original signal with information at discrete points with magnitude of signal present in the original signal at that position. This is illustrated in the figure below.

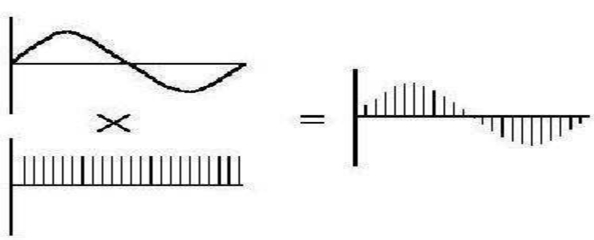

Fig 2 Sampling and Quantization

This can be implemented using an AND gate as shown below.

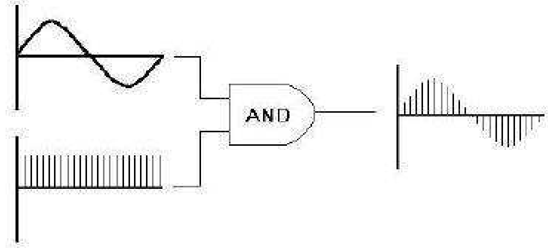

Fig. 3

Sampling and Quantization using an AND gate

Now samples of the input signal are taken at discrete points. But in order to manipulate the signal with a microprocessor or micro controller, these pulses has to be converted in to numbers. Before this conversion, the pulses are quantized to some finite number of quantization levels. For example, if the quantization levels are 0,1,2,3, etc... , a pulse with magnitude between 1 and 2 will be quantized to either 1 or 2. This in fact introduces a noise in the sampled signal. Such noise is known as quantization noise.

III RELATED WORK

In this paper, they considered a fading relay channel operating according to the amplify-and-forward protocol, where each node has full CSI and multiple-antennas are deployed at the relay node. Maximization of the achievable rate with respect to the linear processing at the relay and the power allocation at the source and relay is performed under an instantaneous sum power constraint. In particular, it is assumed that the total power used by all the active nodes during each time slot is fixed. The system under analysis, consists of a source node, a destination node, and a multi-antenna relay node deployed with N antennas, which amplifies the received signal and forwards it towards the destination node. In order to comply with the practical half-duplex constraint, the relay transmission occurs according to a Time Division Duplex (TDD) protocol. In the first time slot the source broadcasts the signal x1 to the relay and the destination. In the second time slot, the relay re-transmits an amplified version of x1, while the source sends a second signal x2, chosen independently from x1. The time durations of the two time-slots are equal. The destination then jointly decodes x1 and x2, given the symbols received in the two time-slots.

The power constraints allow the transmission scheme to encompass and generalize direct transmission and the AF cooperative protocols.In fact,if (H) = 1 the relay is not used at all, neither during the first nor in the second time-slot, and the communication scheme boils down to direct transmission from the source to the relay node; if (H) = 0 only the relay is active during the second timeslot and the communication protocol. Finally, if (H) = 1/2 the communication protocol employs both the relay and the source in the second time slot, and thus resembles the scheme. Notice that, under the considered ergodicity assumption, we could have also enforced a long-term power constraint this modification would complicate the analysis and is not further considered in this work.

Disadvantages:

- Low system throughput.
- High interference between transmitter and receiver subcarriers.
- Referable SINR(Signal to Interference Noise Ratio)

IV. PROPOSED SYSTEM

Quadrature Amplitude Modulation (QAM)

It is both an analog and a digital modulation scheme. It conveys two analog message signals, or two digital bit streams, by changing (modulating) the amplitudes of two carrier waves, using the Amplitude-Shift Keying (ASK) digital modulation scheme or Amplitude Modulation (AM) analog modulation scheme. The two carrier waves, usually sinusoids, are out of phase with each other by 90° and are thus called quadrature carriers or quadrature components — hence the name of the scheme. The modulated waves are summed, and the resulting waveform is a combination of both Phase-Shift Keying (PSK) and Amplitude-Shift Keying (ASK), or (in the analog case) of Phase Modulation (PM) and Amplitude Modulation. In the digital QAM case, a finite number of at least two phases and at least two amplitudes are used. PSK modulators are often designed using the QAM principle, but are not considered as QAM since the amplitude of the modulated carrier signal is constant. QAM is used extensively as a modulation scheme for digital telecommunication systems. Arbitrarily high spectral efficiencies can be achieved with QAM by setting a suitable constellation size, limited only by the noise level and linearity of the communications channel.

Digital QAM

Like all modulation schemes, QAM conveys data by changing some aspect of a carrier signal, or the carrier wave, (usually a sinusoid) in response to a data signal. In the case of QAM, the amplitude of two waves, 90° out-of-phase with each other (in quadrature) are changed (modulated or keyed) to represent the data signal. Amplitude modulating two carriers in quadrature can be equivalently viewed as both amplitude modulating and phase modulating a single carrier.

Phase Modulation (analog PM) and Phase-Shift Keying (digital PSK) can be regarded as a special case of QAM, where the magnitude of the modulating signal is a constant, with only the phase varying. This can also be extended to Frequency Modulation (FM) and Frequency-Shift Keying (FSK), for these can be regarded as a special case of phase modulation.

Analog QAM

When transmitting two signals by modulating them with QAM, the transmitted signal will be of the form:

$$s(t) = R\{[I(t) + iQ(t)]e^{i2\pi f_0 t}\}$$

$$= I(t)\cos(2\pi f_0\, t) - Q(t)\sin(2\pi f_0\, t) \quad (3.1)$$

where $i^2 = -1$, I(t), and Q(t) are the modulating signals, f_0 is the carrier frequency and R{} is the real part.

At the receiver, these two modulating signals can be demodulated using a coherent demodulator. Such a receiver multiplies the received signal separately with both a cosine and sine signal to produce the received estimates of I(t) and Q(t) respectively. Because of the orthogonality property of the carrier signals, it is possible to detect the modulating signals independently.

In the ideal case I(t) is demodulated by multiplying the transmitted signal with a cosine signal:

$$r(t) = s(t)\cos(2\pi f_0 t)$$

$$= I(t)\cos(2\pi f_0\, t)\cos(2\pi f_0\, t) - Q(t)\sin(2\pi f_0\, t)\cos(2\pi f_0\, t) \quad (3.2)$$

Using standard trigonometric identities, we can write it as:

$$r(t) = \frac{1}{2} I(t)\,[1+\cos(4\pi f_0\, t)] - \frac{1}{2}Q(t)\sin(4\pi f_0 t)$$

$$= \frac{1}{2} I(t) + \frac{1}{2}[I(t)\cos(4\pi f_0\, t) - Q(t)\sin 4\pi f_0 t)] \quad (3.3)$$

Low-pass filtering r(t) removes the high frequency terms (containing $4\pi f_0$ t), leaving only the I(t) term. This filtered signal is unaffected by Q(t), showing that the in-phase component can be received independently of the quadrature component. Similarly, we may multiply s(t) by a sine wave and then low-pass filter to extract Q(t).

Orthogonal frequency-division multiplexing (OFDM)

It is a method of encoding digital data on multiple carrier frequencies. OFDM has developed into a popular schemefor wideband digitalcommunication, whether wireless or over copper wires, used in applications such as digital television and audio broadcasting, DSL Internet access, wireless networks, and 4G mobile communications.

The primary advantage of OFDM over single-carrier schemes is its ability to cope with severe channel conditions (for example, attenuation of high frequencies in a long copper wire, narrowband interference and frequency-selective fading due to multipath) without complex equalization filters. Channel equalization is simplified because OFDM may be viewed as using many slowly modulated narrowband signals rather than one rapidly modulated wideband signal. The low symbol rate makes the use of a guard interval between symbols affordable, making it possible to eliminate inter Symbol Interference (ISI) and utilize echoes and time-spreading (on analogue TV these are visible as ghosting and blurring, respectively) to achieve a diversity gain, i.e. a signal-to-noise ratio improvement. This mechanism also facilitates the design of Single Frequency Networks (SFNs), where several adjacent transmitters send the same signal simultaneously at the same frequency, as the signals from multiple distant transmitters may be combined constructively, rather than interfering as would typically occur in a traditional single-carrier system.

Transceiver Architecture

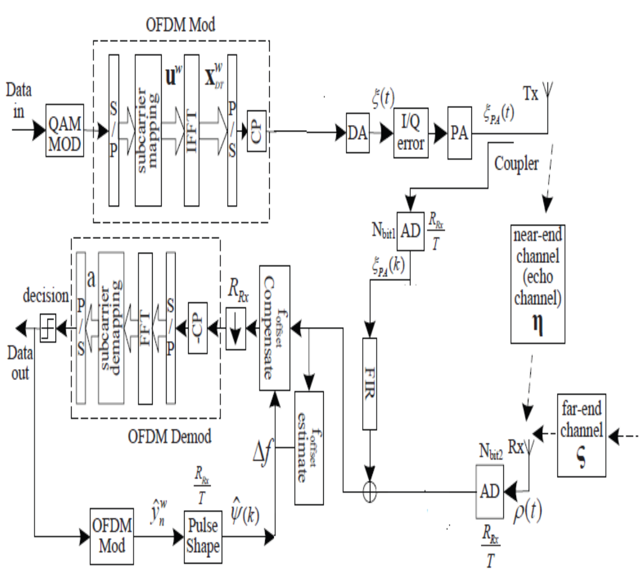

Fig. 4 Transceiver Architecture

To address the effects of the non-ideal subsystem characteristics it is necessary to utilize special transceiver structures. In this work we make use of the transceiver structure as shown in Figure which is based on baseband echo cancellation and has also been used to achieve full duplex communication in wireless systems. For simplicity of illustration, it is denoted in baseband equivalent form. The block labeled “I/Q error” represents the effect of I/Q imbalance introduced during up-conversion. “PA” represents the nonlinear effect of the power amplifier, and the nonlinearity introduced in other steps (e.g., up-conversion) may also be included. This transceiver differs from a conventional one in that baseband echo cancellation is employed at the receiver side before detecting the desired signal. The output signal of the power amplifier is used as a reference signal for echo cancellation (ξPA (k) in Fig.3.2), which is obtained by attaching a coupler to the transmit antenna and using another RF front-end to transform the acquired signal to baseband samples. We utilize a Finite Impulse Responce (FIR) to model the Channel Impulse Response (CIR) between Tx and Rx antennas, whose coefficients are determined by channel estimation using the received signal and the output signal of the power amplifier. A replica of the received near-end signal is then generated and subtracted from the received signal as shown in Fig.3.2. Using this approach allows the spurious signal components caused by transmit I/Q imbalance and PA nonlinearity to be suppressed. Note that in addition to the proposed approach, nonlinear echo cancellation can also be applied in our system. For example, Volterra filter can be employed to model the nonlinear signal path from the transmitter to the receiver.

Subsystem Imperfections

The subsystem imperfections that are important to consider include Carrier Frequency Offset (CFO), time synchronization, quantization error of Analog/Digital Converter (ADC), nonlinearity of Power Amplifier (PA), I/Q imbalance, Phase Noise of Local Oscillator (LO) and Time Jitter of ADC/DAC. Their effects on the performance of subcarrier-based duplexing system are detailed as follows.

Quantization Error

Quantization, in mathematics and digital signal processing, is the process of mapping a large set of input values to a smaller set – such as rounding values to some unit of precision. A device or algorithmic function that performs quantization is called a quantizer. The round-off error introduced by quantization is referred to as quantization error.

In analog-to-digital conversion, the difference between the actual analog value and quantized digital value is called quantization error or quantization distortion. This error is either due to rounding or truncation. The error signal is sometimes considered as an additional random signal called quantization noise because of its stochastic behavior. Quantization is involved to some degree in nearly all digital signals processing, as the process of representing a signal in digital form ordinarily involves rounding. Quantization also forms the core of essentially all lossy compression algorithms.

In subcarrier-based duplexing systems, since the received near-end signal is much stronger than the far-end signal, the ADC at the receiver side needs to have sufficiently high resolution so that the far-end signal is not overwhelmed by the quantization noise1. Assuming uniform quantization is being used, the power of quantization noise can be approximated by

$$P_{QN} = \frac{V_S^2}{3 \times 4^{Nq}}$$

Basic Properties And Types Of Quantization

Because quantization is a many-to-few mapping, it is an inherently non-linear and irreversible process (i.e., because the same output value is shared by multiple input values, it is impossible in general to recover the exact input value when given only the output value).The set of possible input values may be infinitely large, and may possibly be continuous and therefore uncountable (such as the set of all real numbers, or all real numbers within some limited range). The set of possible output values may be finite or countably infinite. The input and output sets involved in quantization can be defined in a rather general way. For example, vector quantization is the application of quantization to multi-dimensional (vector-valued) input data.There are two substantially different classes of applications where quantization is used:

The first type, which may simply be called rounding quantization, is the one employed for many applications, to enable the use of a simple approximate representation for some quantity that is to be measured and used in other calculations. This category includes the simple rounding approximations used in everyday arithmetic. This category also includes analog-to-digital conversion of a signal for a digital signal processing system (e.g., using a sound card of a personal computer to capture an audio signal) and the calculations performed within most digital filtering processes. Here the purpose is primarily to retain as much signal fidelity as possible while eliminating unnecessary precision and keeping the dynamic range of the signal within practical limits (to avoid signal clipping or arithmetic overflow). In such uses, substantial loss of signal fidelity is often unacceptable, and the design often centers around managing the approximation error to ensure that very little distortion is introduced.

The second type, which can be called rate–distortion optimized quantization, is encountered in source coding for "lossy" data compression algorithms, where the purpose is to manage distortion within the limits of the bit rate supported by a communication channel or storage medium. In this second setting, the amount of introduced distortion may be managed carefully by sophisticated techniques, and introducing some significant amount of distortion may be unavoidable. A quantizer designed for this purpose may be quite different and more elaborate in design than an ordinary rounding operation. It is in this domain that substantial rate–distortion theory analysis is likely to be applied. However, the same concepts actually apply in both use cases. The analysis of quantization involves studying the amount of data (typically measured in digits or bits or bit rate) that is used to represent the output of the quantizer, and studying the loss of precision that is introduced by the quantization process (which is referred to as the distortion). The general field of such study of rate and distortion is known as rate–distortion theory.

Scalar Quantization

The most common type of quantization is known as scalar quantization. Scalar quantization, typically denoted as y=Q(x), is the process of using a quantization function Q( ) to map a scalar (one-dimensional) input value x to a scalar output value y. Scalar quantization can be as simple and intuitive as rounding high-precision numbers to the nearest integer, or to the nearest multiple of some other unit of precision (such as rounding a large monetary amount to the nearest thousand dollars). Scalar quantization of continuous-valued input data that is performed by an electronic sensor is referred to as analog-to-digital conversion. Analog-to-digital conversion often also involves sampling the signal periodically in time (e.g., at 44.1 kHz for CD-quality audio signals).

I/Q Imbalance

The In-phase (I) and the Quadrature (Q) channels are necessary for any angle modulated signals because the two sidebands of the RF spectrum contain different information and may result in irreversible corruption if they overlap each other without being separated into two phases.The demodulator at the receiver has to be synchronous in nature. The receiver must possess an oscillator, which is at the exactly same frequency, and phase as the carrier oscillator, at the transmitter end. The low pass filters at the two paths of the receiver must have identical characteristics, as any mismatch in their characteristics would lead to I/Q errors. The effect of I/Q imbalance is modeled as $s' = \alpha s + \beta s^*$, where s represents the original signal, is the resulting signal and α and β are constants. The term βs^* is a “mirror image” of the original signal and will cause ICI. Its power can be written as

$$P_{s,IQ} = \gamma IQPs$$

LO Phase Noise

All super heterodyne receivers use one or more local oscillators to convert an input frequency to an intermediate frequency before the signal is demodulated. In the ideal receiver, these frequency conversions would not distort the input signal, and all information on the signal could be recovered. In a real-world receiver, both the mixer used for converting the signal’s frequency and the local oscillator will distort the signal and limit the receiver’s ability to recover the modulation on a signal. Mixer degradations, such as undesired mixing products, can be minimized by proper design in the rest of the receiver. The local oscillator degradations, which are principally random phase variations known as phase noise, cannot be decreased except by improving the performance of the oscillator.

Local oscillator phase noise is a necessity for many receiving systems. The local oscillator phase noise will limit the ultimate signal-to-noise ratio which can be achieved when listening to a Frequency Modulated (FM) or Phase-Modulated (PM) signal. The performance of some types of amplitude modulation detectors may be degraded by the local oscillator phase noise. When the receiver is used to monitor Phase-Shift Keyed (PSK) or Frequency-Shift Keyed (FSK) signals, the phase noise may limit the maximum bit error rate which the system can achieve. In FM/FDM (Frequency Division Multiplex) systems, phase noise will often limit the maximum noise power ratio of the receiving system. Phase noise can limit the maximum angular resolution which can be achieved by an interferometric direction-finding receiver. Reciprocal mixing may cause the receiver noise floor to increase when strong signals are near the receiver’s tuned frequency; this limits the ability to recover weak signals. All of these effects are due to local oscillator phase noise, and can only be reduced by decreasing the phase noise of the oscillator.

Time Jitter of ADC/DAC

Jitter is the variation in the time of an event—such as a regular clock signal—from nominal. For example, the jitter on a regular clock signal is the difference between the actual pulse transition times of the real clock and the transition times that would have occurred had the clock been ideal, that is to say, perfectly regular. Against this nominal reference, the zero-crossing transitions of many of the pulses in a jittered data stream are seen to vary in time from the ideal clock timing. Expressed another way, jitter is phase modulation of the digital interface signal. The jitter component can be extracted from the clock or digital interface signal to be analyzed as a signal in its own right. Among the more useful ways of characterizing jitter is by examining its frequency spectrum and identifying the significant frequency components of the jitter itself.When very little jitter is present, the pulse transitions are moved back or forth by only small measures of time. When the jitter is increased, the transitions move across a larger range of times.Jitter amplitude, then, is a measure of time displacement and is expressed in units of time, either as fractions of a second or unit intervals. For those new to jitter measurement, this can lead to some disconcerting graph labels, with time on the vertical axis versus time on the horizontal axis, for example.Jitter frequency is the rate at which this phase-shifting is taking place. Like other noise or interference signals, the jitter modulation signal can be a pure and regular sine wave, a complex waveform or have a completely random character.

The AES3 digital audio interface format1 now has specifications for jitter. (The consumer version of the interface, which is described in IEC60958-3:20002 also have jitter specifications.) This specification was drawn up to resolve problems that would occur when units that conformed to the interface specification were interconnected and yet the interface did not work reliably.

Time Synchronization

There are two sets of synchronization problems in OFDM, frequency error and timing error. The frequency error is caused by mismatch between the transmitter and receiver oscillators and also the channel Doppler shift. And the timing error is caused by symbol timing and sample timing which is mostly caused by clock drift. Fig. shows the diagram of synchronization errors, the things we have detected in our project are symbol timing and frequency offset which are colored in green.

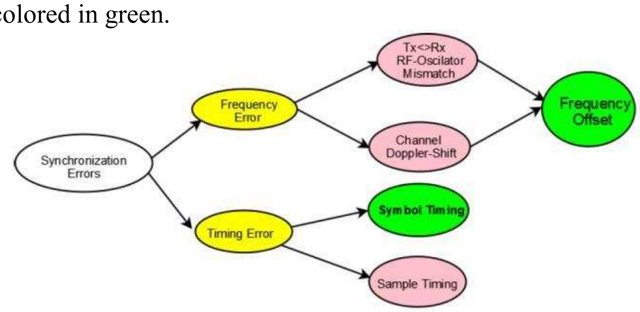

Fig. 5 Synchronization Errors

Synchronization Techniques

There are three major synchronization techniques described as below.

- Pilot Symbols: here we use preamble in OFDM symbols. Its drawback is that lowers the achievable data rate.
- Blind Synchronization Techniques: here we use cyclo-stationarity in OFDM signal. Its drawback is that is costly in computation.
- Cyclic-Prefix Based Techniques: this method is efficient as we don't use pilot symbols and any way we need cyclic prefix for removing the multipath effect so they are freely available to be used for synchronization. In the next part we explain the cyclic-prefix method.

Cyclic-Prefix Based Techniques

Cyclic Prefix is a copy of last part of OFDM symbol to the beginning of the symbol and is removed at the receiver before demodulation. It should be at least as long as significant part of impulse response experienced by the transmitted signal. In the following Fig.3.4 Ncp is the number of cyclic prefix and N is number of samples. So one of our tasks in this project is to find the symbol start in OFDM data stream.

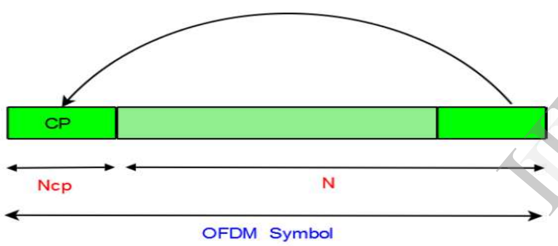

Fig. 6 OFDM Symbol Structure

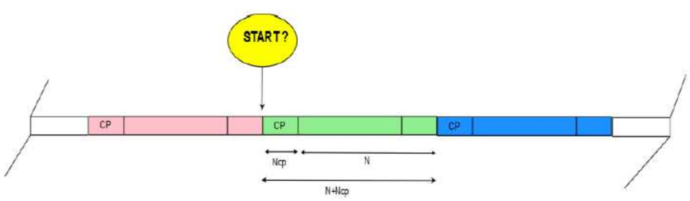

Fig. 7 Symbol Start in OFDM Data Stream

If the symbol start becomes wrong, it will lead to timing errors. Then we will lose the orthogonality of the OFDM signal and have Inter Symbol Interference (ISI) and phase error. We need to use some synchronization techniques to find the correct symbol start and carrier frequency offset. There are two OFDM CP-based synchronization techniques which are implemented in our project.

Echo Cancellation

The term echo cancellation is used in telephony to describe the process of removing echo from a voice communication in order to improve voice quality on a telephone call. In addition to improving subjective quality, this process increases the capacity achieved through silence suppression by preventing echo from traveling across a network. Echo cancellation involves first recognizing the originally transmitted signal that re-appears, with some delay, in the transmitted or received signal. Once the echo is recognized, it can be removed by 'subtracting' it from the transmitted or received signal. This technique is generally implemented using a Digital Signal Processor (DSP), but can also be implemented in software.

Baseband echo cancellation is carried out on a frame basis and its procedure is described as follows. Consider a node operating in subcarrier-based duplexing mode, we use STx and SRx to denote the set of transmitting subcarriers and receiving subcarriers, respectively (STx and SRx are mutually exclusive). Suppose there are in total W OFDM symbols in each frame. For the w-th OFDM symbol, we use uw = [uw1, uw2, ..., uw,N]T to denote the vector of frequency domain samples to transmit, where uwn is the data symbol to be transmitted on the n-th subcarrier, and N is the length of the Inverse Fast Fourier Transform (IFFT), i.e., the total number of subcarriers. Note that uwn is non-zero only if n E STx. The time-domain samples xw DT = [xw1, xw2, ..., xw,N]T are obtained by applying IFFT to uw, i.e., xw DT = FHuw, where the (m, n)-th entry of F is

$$F_{[m,n]}=\frac{1}{\sqrt{N}}e^{-j\frac{2\pi}{N}(m-1)(n-1)} \quad m,n = 1,2,\ldots\ldots, N$$

.

V. EXPERIMENTAL RESULTS AND DISCUSSION

Performance Of The Proposed Transceiver With Interleaved Subcarrier Allocation.

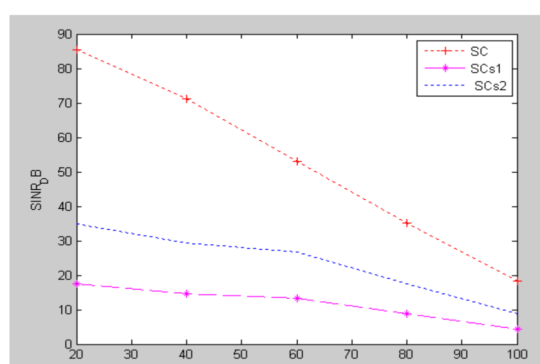

Fig. 8 Performance Of The Proposed Transceiver

Fig. 8 shows the system performance when the subcarriers assigned for transmission and reception are interleaved, i.e., the odd subcarriers (1,3,...,51) are for transmission and even subcarriers (2,4,...,52) are for reception or vice versa.

Total Power Consumption Of 2-User Cooperative OFDMA System For Different Data Rates.

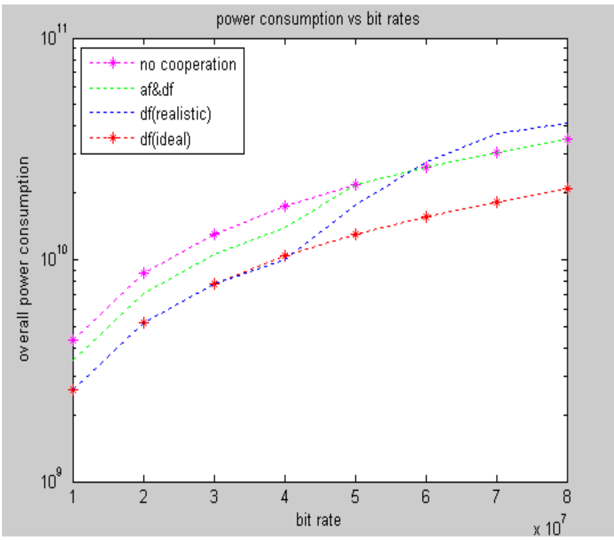

Fig. 9 Total Power Consumption For Different Data Rates

Fig. 9 shows the results when the data rate of user 1 and user 2 varies while *d*10 is fixed to be 50m. It can be seen that the difference between the power consumption of DF cooperation in the ideal case and that in the realistic case increases with the data rate. The reason is that the transmit power of user 1 increases with the data rate, therefore more interference is generated to the data streams from user 2 to user 1, and user 2 needs to scale up its transmit power in order to compensate for the SINR loss. As the data rate increases, the extra transmit power required by user 2 also increases, thus the total power consumption of optimal DF cooperation will finally exceed that of AF&DF cooperation and that of no cooperation. However, it can be seen from Fig.4.2 that optimal DF cooperation is advantageous in most of the data rate region.

Performance Of The Proposed Transceiver For Different ADC Resolutions (Interleaved Subcarrier Assignment).

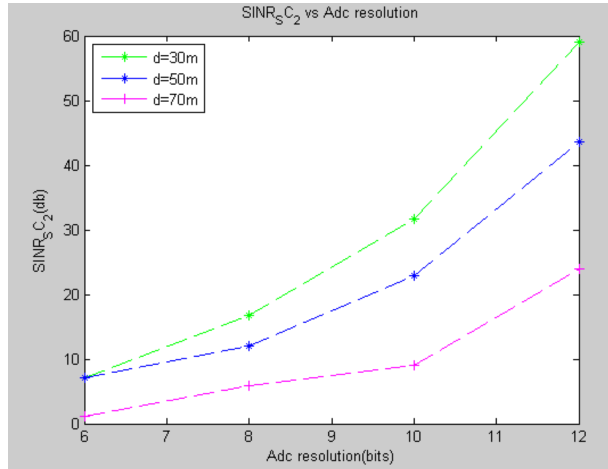

Fig. 10 SINR of the Second Stage Echo Cancellation

Fig.10 shows the per subcarrier SINR of the second stage echo cancellation under different ADC resolutions. It can be noticed that SINRsC2, increases almost linearly with the resolution of ADC when the ADC resolution is less than 12bit, and 1bit increment of ADC resolution improves SINRsC2 by about 6dB, which coincides with the analysis of quantization noise. However, as the resolution of ADC keeps increasing, this improvement become less significant. This is because when the ADC resolution is low, the quantization error dominates the effects of other subsystem imperfections and determines the output SINR; as the precision of ADCs increases, the power of quantization noise decreases and the effects of other subsystem imperfections will emerge.

Performance Of The Proposed Transceiver For Different Degrees Of Pa Nonlinearity (Interleaved Subcarrier Assignment).

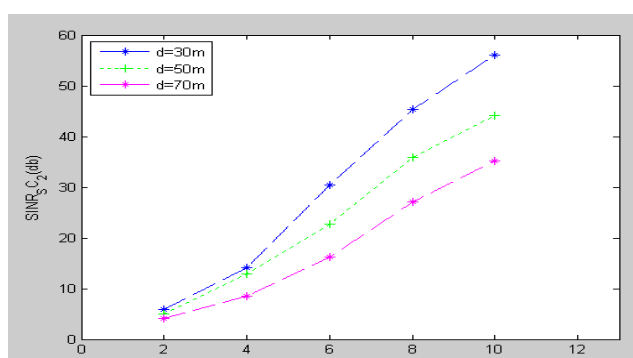

Fig 11 Different Degrees Of PA Nonlinearity

Fig. 11 shows the performance of the proposed transceiver under different degrees of PA nonlinearity. In contrast to quantization error and phase noise, the effect of PA amplifier on SINRsC2 is insignificant. Especially, when the OBO of power amplifier is set to 3dB (which renders γNL equal to 0.02), the per-subcarrier SINR is still satisfactory. The reason is that the nonlinear signal components resulting from PA nonlinearity are cancelled at the receiver side, thus the power of the residue is small compared with those of quantization error and phase noise. Similar observation can be made for the effect of I/Q imbalance, which is omitted here for brevity. Therefore, we can conclude that the effects of quantization error and LO phase noise on SINRSC,s2 are more significant than those of PA nonlinearity and IQ imbalance.

VI. CONCLUSION AND FUTURE WORK

A particular subcarrier resource allocation approach investigated in this paper is a method based on nodes that transmit and receive on adjacent OFDM subcarriers simultaneously. To perform the investigation we proposed a transceiver structure that allows OFDM users to transmit and receive simultaneously on adjacent subcarriers so that the system tradeoffs and limitations of this approach could be understood. The performance of the transceiver was evaluated by both analysis and computer simulation and it was shown that the non-ideal characteristics of subsystems will limit the achievable SINR. This scheme was then also incorporated into a recently proposed cooperation strategy to investigate its performance under realistic conditions. This paper is concentrated only on SINR and echo cancellation… and it is found to be efficient with increased distance. An important parameter of a communication system is BER (Bit Error Rate). Decided to work on BER and to find performance of BER with increased Signal to Noise ratio and distance.

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