Software defined radio based on the upper audio band for low data rate communications over short distances

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***Abstract*—For communicating short data sequences over small distances, the use of devices with conventional RF interfaces requires standardized hardware, dedicated infrastructure and appropriate Link/Network layer protocols. To address challenges associated with these requirements, a communication mechanism using devices which support simple audio interfaces (speakers and microphones) is proposed using the upper audio band (UAB) of frequencies (16-20KHz). Devices with audio interfaces can be deployed in a personal area network for communicating at low data rates over small distances. Multi-tone FSK modulation is used for transmitting Reed-Solomon encoded data over the UAB. For peer-to-peer communication applications, a sensing mechanism is enabled on the receiving device to sense for empty time-frequency slots and schedule its data transmission at the appropriate times. A system prototype is developed using portable speakers and smartphones with sensitive microphones. The effective throughput of the modem is evaluated for different sensing durations and distances. Ad-hoc peer-to-peer networks can be enabled between mobile devices for communicating short data sequences.**

Keywords-Multi-tone FSK, Acoustic band radio, Sensing architecture

I. INTRODUCTION

Conventional Near Field Communication (NFC) enabled devices operate in a Reader/writer mode, Card emulation mode or a Peer-to peer mode. An NFC enabled handset consists of an NFC chipset, a Host/baseband controller, an Antenna and a secure element, which interfaces with the NFC chipset and the Baseband controller. An NFC deployment needs a close interaction between mobile network operators, handset manufacturers and service providers for peer-to-peer applications. The technical challenges involved are a high diversity of handsets with different operating systems, and the need for compliance to standards for protocols, interfaces and hardware.

To address these challenges, we propose to use the conventional audio interfaces available on mobile devices to communicate data for peer-to-peer applications. Speakers and microphones on currently available mobile devices can process frequencies upto 20KHz. Since the frequencies in the band 16-20KHz are almost imperceptible to the human ear, this band can be used to communicate data between devices over the air. In addition, specialized chipsets and associated communication protocols are not needed. Appropriate modulation and error-control coding schemes were implemented to achieve meaningful data rates within the limited bandwidth. Furthermore, since the intended audio spectrum may be occupied by signals from entertainment appliances such as Hi-Fi audio systems, the communication modem was made spectrally agile with appropriate sensing and spectrum access mechanisms.

The audio modem implementing encoding/decoding and modulation is implementable in software, and can be modified/reconfigured according to data rate and performance requirements. This modem is independent of the hardware used in the user handset/mobile device. Conventional microphones with good sensitivity levels are appropriate for receiving and processing the audio signals.

Prior work in [1-3] has used the audio band for communicating data. In [2], multi-carrier modulation has been proposed for transmission of data over the air using the perceptible audio band of 6.4-8KHz. The frequencies in this band are first filtered out from the original sound. A multi-carrier modulation with 32 sub-carriers is then used to communicate data over this band. In [3], data is embedded in a sound signal over the perceptible audio band using a modified Lapped transform on an audio signal. In addition to the inevitable audio signal distortion by these methods, these audio communication systems do not address sensing and access mechanisms for deployment in a noisy environment. Hence there is a higher probability of performance degradation due to interference from sound in the audio band.

The audio band enabled devices can be deployed in a peer-to-peer or a master-slave mode in a closed room environment. Since the use of the audio spectrum by the mobile devices and other audio systems is unpredictable, there is a need to jointly design the spectrum sensing and access mechanisms. In this work, we describe sensing and access mechanisms for devices communicating data over the upper audio band. The data transmission needs to be appropriately scheduled in time and frequency with modulation and coding schemes suited for the data rate and performance requirements. The UAB radio has been designed with two essential components: a) Audio interface management. b) Reconfiguration management. The audio interface management component implements a real-time control of power and bandwidth efficient modulation and error control coding schemes, depending on the data rate and error-rate performance requirements. The reconfiguration management component can reconfigure the spectrum sensing scheduling and channel usage.

To design a reconfigurable audio modem, we define the following:

1. A system model defining the deployment architecture for users in the network
2. A channel model defining the channel usage pattern for the deployed users for a given application.
3. Data Communication mechanisms using appropriate modulation and error control coding schemes.
4. Co-existence mechanisms defining appropriate spectrum sensing methods and spectrum access schemes.

Based on the system and channel models, a functional architecture for management and control of the reconfigurable audio modem is proposed to enable data communication and co-existence with multiple users for medium access of common channels. The specific tasks required for medium access include: a) Sensing scheduling and b) Spectrum aware access control.

A joint design of the sensing and access mechanism is proposed using:

a) A spectrum sensor, which identifies a spectral opportunity.

b) A sensing strategy which decides which channels in the spectrum to sense and

c) An access strategy, which determines the usage pattern for accessing a channel identified by the sensing strategy.

The paper outline is as follows. Section II describes the system deployment model and the user co-operation modes. Section III defines a channel usage model from a medium access perspective. Section IV describes the data communication mechanism using a multi-tone FSK modulation for Reed Solomon encoded data and spectrum sensing mechanisms for the audio spectrum and access strategies. Section V describes the implementation of the reconfigurable modem. Section VI concludes the paper.

II. SYSTEM DEPLOYMENT MODEL

There are two modes of operation possible for deployment of UAB based devices:

1. Peer-to-peer mode- Logical groups are formed around each device and require a contention free data frame exchange between them.

2. Master-slave mode-Under the co-ordination of a Master, logical groups are formed with a Master-Slave operation to facilitate a contention free data frame exchange.

The two non-co-operating users are the Primary users (PU) and Secondary users (SU). The primary users belong to a peer-to-peer network or a master slave network on pre-assigned channels. The PU may also be high fidelity audio systems. Secondary users are new users who want to initiate communication with peers or a master node. The SU uses a spectrum sensor to determine whether the PU is idle or busy at a given time. The UAB spectrum can be used in two modes:

a) A broadcast mode, where a master device broadcasts information over the UAB to users in its group.

b) A peer-to-peer mode where all deployed devices have microphones which can be used for sensing the UAB and can schedule their data transmission by using appropriate time-frequency slots.

III. CHANNEL USAGE MODEL

The PU traffic is assumed to be slotted. The channel usage pattern of PUs is not available *apriori* and needs to be sensed. The UAB devices need to periodically schedule their sensing operations to access the available Time-Frequency slots. A medium access mechanism for a user to operate in a peer-to-peer or master slave mode using appropriate frequency bands and time slots is needed. This requires defining a sensing strategy and an access strategy. A sensing strategy determines which channel to sense. An access strategy determines the usage pattern of a sensed/usable channel.

There are two use cases, which determine the sensing strategy for a UAB device:

1. The device cannot access all of the available channels simultaneously- In this case, the sensing strategy needs to choose which channels each user should attempt to use in different time slots, in order to optimally utilize the spectral opportunities. A study on the optimal method for choosing channels to sense using an order optimal single index strategy is done in [4].
2. The device can access all the channels simultaneously- In this case, the sensing strategy senses for the presence of all the channels simultaneously.

IV. DATA COMMUNICATION MECHANISM

A. Channel Characteristics

The audio channel over the air can be modeled as a linear time invariant system. The noise was experimentally observed to have a Gaussian distribution. These channel characteristics can be used as guidelines for defining the signal model and the optimal receiver.

B. *Signal Model*

Signals which can pass undistorted over a linear time invariant channel can be modeled as eigenfunctions. Multi-tone FSK modulated signals are eigenfunctions which are suited for the audio channel. From a spectrum sensing perspective, since sampling and processing a broadband spectrum is difficult for devices with audio interfaces, multi-tone FSK signals with spectral lines are easier to sense. A 2-tone FSK signal is given by:

$$x(t) = \left[A\cos(2\pi f_1 t) + A\cos(2\pi f_2 t) \right] \quad 0 < t \le T \tag{1}$$

The noisy received signal sampled at a default rate of $f_s = 44100$ samples/s is given by:

$$y(nt_s) = \left[A\cos(2\pi f_1 n t_s) + A\cos(2\pi f_2 n t_s) \right] + w(nt_s)$$
$$0 \le n \le N-1 \quad t_s = 1/f_s \tag{2}$$

In the presence of Additive White Gaussian Noise (AWGN) $w(n\,t_s)$, the estimates of the frequencies in (2) are obtained by peak-picking the Periodogram of the received signal. The joint probability density function (p.d.f) of two frequency estimates is a Gaussian with a circular contour plot. It is known that a dense packing of circles in a 2 dimensional plane is a Hexagonal lattice packing. If the frequencies $[f_1, f_2]$ are chosen from a Hexagonal lattice, the Voronoi region for each lattice co-ordinate point is a hexagon. The generator matrix for a hexagonal lattice is given by [5]:

$$G = \begin{bmatrix} 1 & 0 \\ 0.5 & \sqrt{3}/2 \end{bmatrix} \tag{3}$$

With a base pair of frequencies, M1=13000 Hz, M0=12000 Hz and a translation of $T = [0, 6500]$ Hz, we compute the 4 pairs of tones for a 2-tone FSK modulation to obtain the set of 4 adjacent frequency lattice co-ordinates $F = [F_1, F_2, F_3, F_4]$ where:

$F_1 = [f_{11}, f_{12}]$ = [M0 M0] G +[T]= [18000 Hz,16892 Hz]

$F_2 = [f_{21}, f_{22}]$ = [M0 M1] G +[T]= [18500 Hz,17758 Hz]

$F_3 = [f_{31}, f_{32}]$ = [M1 M0] G +[T]= [19000 Hz,16892 Hz]

$F_4 = [f_{41}, f_{42}]$ = [M1 M1] G +[T]= [19500 Hz,17758 Hz]

(4)

The spacing between adjacent co-ordinate points in the lattice is 1000 Hz. This spacing can be reduced by reducing the difference between M1 and M0. A reduced spacing between co-ordinate points will allow more channels to be introduced. A channel is defined as the frequency band spanned by a set of 4 adjacent lattice co-ordinate points. Multiple channels can be introduced by changing the lattice translation vector *T*.

The signal model can be extended to a 3-tone FSK modulation where the three tones are chosen from a laminated 3 dimensional Hexagonal lattice. The generator matrix for a 3 dimensional laminated hexagonal lattice can be derived easily. A sphere packing is called a lattice packing if it has the property that 0 is a center and that if there are spheres with centers u, v then there are spheres with center $u+v, u-v$. We can find n centers $v_1, v_2, ..., v_n$ for an n-dimensional lattice such that the set of all centers is given by the sum $\sum_i k_i v_i$ where k_i are integers. The vectors $v_1, v_2, ..., v_n$ are then called the basis for the lattice. As a first step, spheres are packed in a Hexagonal lattice structure in the 2 dimensional plane. Next, a new layer of spheres is placed with their centers on top of the deep holes of the 2 dimensional Hexagonal lattice packing. A unit vector in the direction of the center of the second layer of spheres forms a part of the generator matrix for this laminated 3 dimensional Hexagonal lattice. The generator matrix is given by:

$$G = \begin{bmatrix} 1 & 0 & 0 \\ 0.5 & \sqrt{3}/2 & 0 \\ 0.5 & 0.5/\sqrt{3} & \sqrt{2/3} \end{bmatrix} \tag{5}$$

This generator matrix can be used to derive frequencies in the range of 16-20KHz, which can be used for a 3-tone FSK signal.

C. *Spectrum Sensing*

UAB based devices need to sense the audio spectrum so that they can schedule their data transmission on appropriate channels without causing interference. To detect the presence of frequencies in the 2-tone FSK signal in (4), an un-windowed periodogram is used to compute a test statistic given by [6]:

$$r_{k1} = \frac{2N.S(k_1)}{\sum_{k=0}^{N-1} S(k)} \quad \text{and} \quad r_{k2} = \frac{2N.S(k_2)}{\sum_{k=0}^{N-1} S(k)} \quad \text{for}$$
$$k_1 = \frac{N.f_{i1}}{f_s},\; k_2 = \frac{N.f_{i2}}{f_s} \quad i = 1,2,3,4 \tag{6}$$

The periodogram $S(k)$ is computed using an N point FFT over the received signal $y(n)$ as:

$$S(k) = \frac{1}{N}\left| \sum_{n=0}^{N-1} y(n) \exp\left(\frac{-j\,2\pi\, n\, k}{N} \right) \right|^2$$
$$k = 0,1,,, N-1 \tag{7}$$

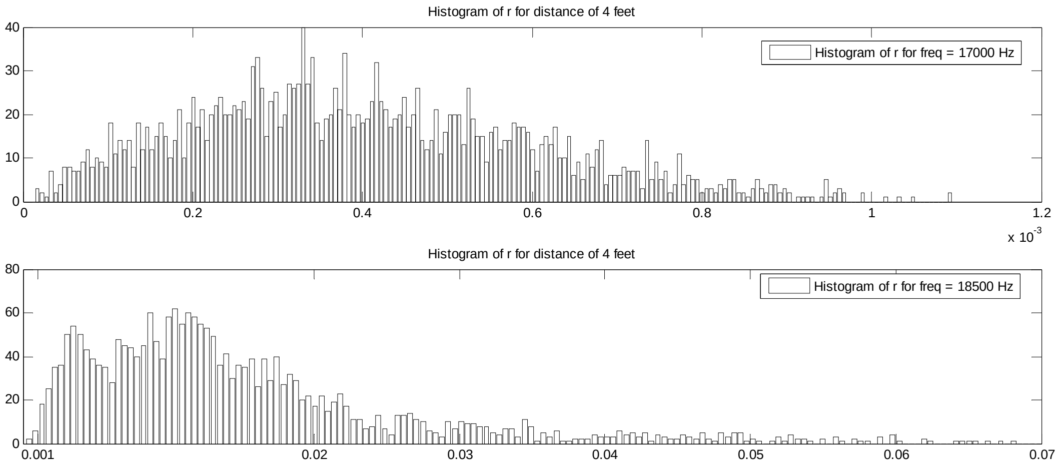

Figure 1. Histogram plots of the test statistic for tone detection.

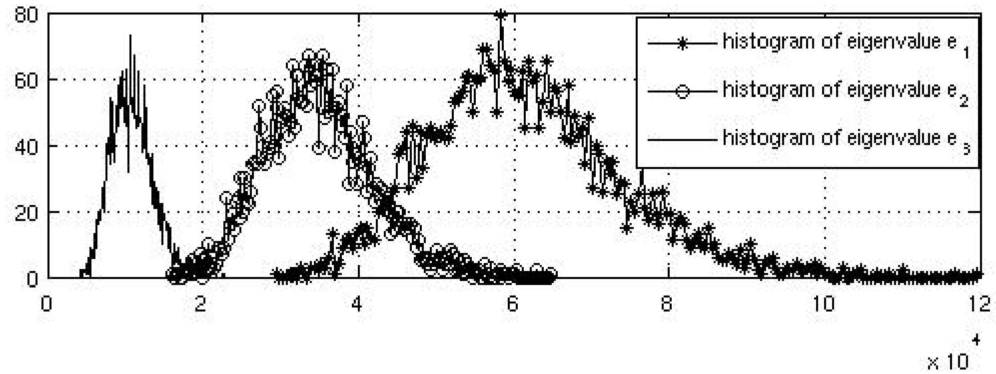

Figure 2. Histogram of the eigenvalues of the STFT covariance matrix for SNR = -8dB

For a noisy received signal, the periodogram magnitudes at the tone pairs in the set F have a non-central Chi-Squared distribution for which the test statistic r_k has been shown to be optimal for tone detection [6]. A histogram plot of the test statistic r_k in the presence of a tone ($f = 18500\,Hz$) from the frequency pairs in (4) is shown in Fig 1. The histogram plot for the test statistic r_k evaluated for a frequency not in the set F ($f = 17000\,Hz$) is shown for comparison. The sensing duration was taken as 20ms. The distance between the speaker and microphone was 4 feet. It can be seen that a threshold of 10^{-3} can be used on the test statistic to detect the presence/absence of valid tones in the received signal. It was experimentally observed that the threshold for the test statistic remained approximately the same for a distance of 2-6 ft between the speaker and the microphone.

The sensing for detection of valid tones can be made more sensitive for a smaller sensing duration. In [7], a covariance matrix of the Short Time Fourier Transform (STFT) of the received signal is computed and a ratio of the sum of diagonal elements to the sum of off-diagonal elements in the covariance matrix is taken to detect the presence of a 'white space'. In [8], a ratio of the maximum and minimum eigenvalues of the covariance matrix of the received signal is taken to detect the presence of a signal. In this paper, we compute the covariance matrix of the STFT of the multi-tone FSK signal. The two major eigenvalues indicate the presence of two tones in the 2-tone FSK signal. In the presence of additive white Gaussian noise, the eigenvalues were observed to have a Gaussian distribution. A threshold is chosen to differentiate between the two major eigenvalues and the smaller eigenvalues. At the receiver, an N point FFT is used to compute the STFT of the received noisy signal:

$$S_\tau[k] = \frac{1}{N}\left|\sum_{n=0}^{N-1} y[n+\tau N]\, e^{-j\frac{2\pi nk}{N}}\right|^2 \qquad (8)$$

Where $\tau \in \{0,1,...,N_s - 1\}$ is the index of the sensing window with the number of sensing windows N_s.

Let $\mathbf{s}_i \quad i = 1,...,N$ denote the spectrograms each stretched out into column vectors. Assume that the spectrograms are of length F. The spectrograms are arranged as a matrix S of size $F \times N$. An unbiased estimate of the covariance matrix of size $F \times F$ is estimated using:

$$C_K = \frac{1}{K-1} \sum_{k=1}^{K} S_k S_k^T \qquad (9)$$

for K time intervals. For a two-tone FSK signal, the two largest eigenvalues of the covariance matrix are computed and compared to a threshold. The two largest eigenvalues correspond to the two tones in the received signal. The threshold is computed by taking the intersection of the histogram of the two largest eigenvalues and the histogram of the eigenvalues with smaller magnitudes. For a three tone FSK signal, the three largest eigenvalues can be chosen for comparing against a threshold. In Fig 2, a histogram plot of the two largest eigenvalues e_1, e_2 is compared with the histogram of a third eigenvalue e_3. The distributions are approximately Gaussian. A time duration of 2.5ms was taken to compute the STFT using a 512 point FFT and the covariance matrix for a SNR= -8dB. Thresholds chosen appropriately on the two largest eigenvalues will enable the modem to detect the presence/absence of valid tones in the received signal.

D. Receiver

Once a valid channel is identified based on the sensing results, the receiving device proceeds to the demodulation and decoding stages. A maximum likelihood estimate of the frequencies $\hat{\mathbf{f}} = [\hat{f}_1, \hat{f}_2]$ in the received two-tone FSK signal is obtained by choosing the two frequencies at the two largest peaks of the Periodogram of the received signal. For all the frequency pairs in (4),

$$|f_{i1} - f_{i2}| / f_s >> \frac{1}{N} \quad i = 1,2,3,4 \qquad (10)$$

Under this condition, the components of the vector of frequency estimates $\hat{\mathbf{f}} = [\hat{f}_1, \hat{f}_2]$ are uncorrelated and the joint p.d.f of the vector of frequency estimates is Gaussian and has a circular contour plot. Hence, using the minimum Euclidean distance as a decoding metric will maximize the likelihood of correct frequency decoding. The decoded frequency pair is given by:

$$[f_1^{dec}, f_2^{dec}] = \min_{\{f_{k1}, f_{k2}\}} \left[\left| \hat{f}_1 - f_{k1} \right|^2 + \left| \hat{f}_2 - f_{k2} \right|^2 \right] \qquad (11)$$
$$k = 1,2,3,4$$

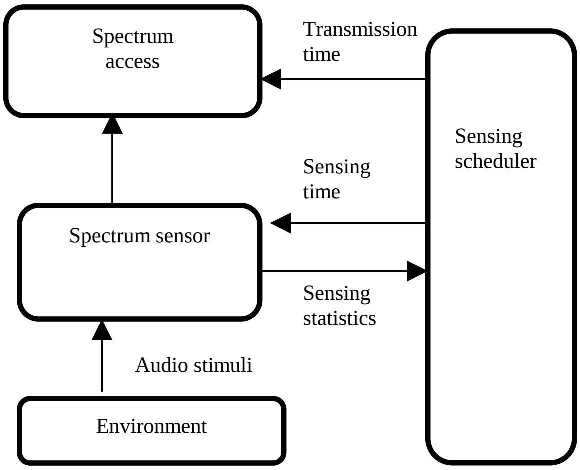

Figure 3. Spectrum sensing functions

The maximum likelihood estimate of the frequencies, $[f_1^{dec}, f_2^{dec}]$ are mapped back to the corresponding data symbols. The error rate performance of a 2-tone and 3-tone FSK modulation over an AWGN channel has been evaluated using Monte Carlo simulation and is reported in [9]. A significant gain in error rate performance was observed compared to conventional 4/8-ary FSK.

V. SPECTRUM SENSING AND ACCESS FUNCTIONS

The Physical layer functions relevant to the medium access mechanism are a) spectrum sensing, b) rate adaptation and c) error control. The audio spectrum can be sensed periodically. The UAB modem is capable of changing its modulation schemes from a 2-tone to a 3-tone FSK modulation to provide different data rates. Based on the sensing and access strategies, the throughput of a device can be optimized by adapting the modulation. The essential components of a spectrum sensing audio modem are shown in Fig 3. The functionality of the components are as follows [10, 11]:

1. Spectrum sensor: The audio spectrum is sensed by the microphone for the presence/absence of usable channels.
2. Sensing scheduler: This module regulates the sensing duration and data transmission times. The sensing scheduler works in conjunction with the spectrum-sensing block.
3. Spectrum access function: A spectrum access strategy in this module enables multiple users to share a common channel by determining who will access the channel, or when a user accesses the channel.

Based on the throughput and performance requirements, the sensing time needs to be minimized and the transmission time needs to be maximized. The sensing statistics are an estimate of the thresholds, which are used on test statistics for spectrum sensing.

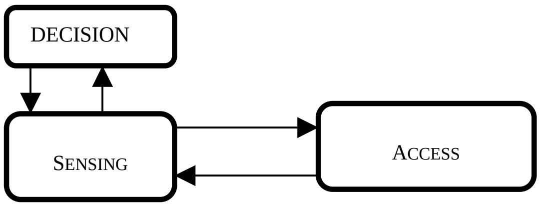

Figure 4. Operational stages of the medium access mechanism

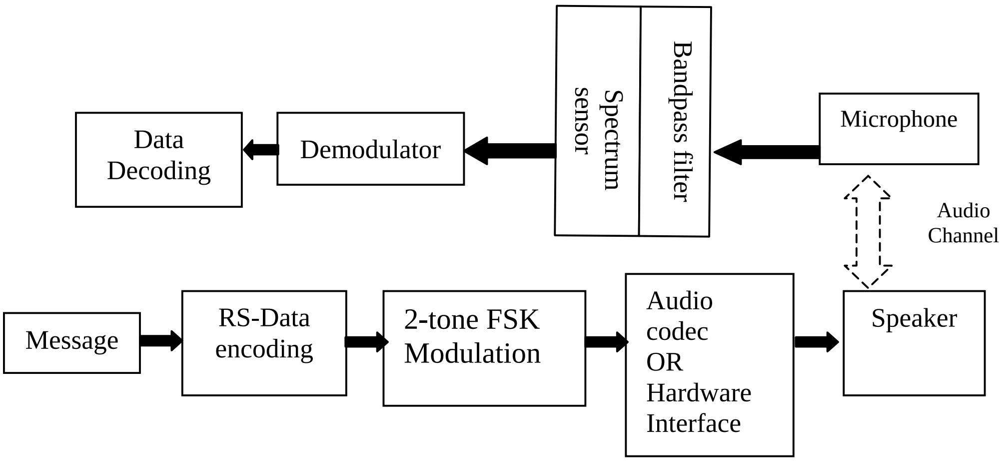

Figure 5. Transmit/Receive chain for the UAB modem

The sensing scheduler schedules the sensing time for the spectrum sensor and transmission time for the spectrum access block. For the channel model and the sensing strategy described in section III, an appropriate access strategy to maximize the throughput is needed. A State diagram describing the operational stages for the medium access mechanism is shown in Fig 4. The operational stages of the Medium access mechanism can be decomposed as follows [12]:

1. Decision stage-The SU transmitter decides which channels to sense based on the sensing strategy.
2. Sensing stage-The SU transmitter senses the selected channels.
3. Access stage-If the access channel is sensed to be free, a data frame is transmitted by the secondary user.

VI. MODEM PROTOTYPE

A. Modem Architecture

A prototype reconfigurable modem was developed using the upper audio band of frequencies and the proposed multi-tone modulation format. The modem was designed to operate on two channels. For data transmission, a message sequence $D = [d_1, d_2, ...d_{32}]$ with $d_i \in \{0,1\}$ is appended with a flag sequence $[s_1, s_2, s_3, s_4]$, giving a 36 bit sequence $[D,[s_1, s_2, s_3, s_4]]$ which is encoded using a three symbol error correcting (15,9) Reed Solomon (RS) code over the field GF(2^4). The field elements of the RS code over the field GF(2^4) are generated using a primitive polynomial $p(X) = 1 + X + X^4$. The encoded symbols are mapped to a 2-tone FSK signal for data transmission over the UAB. Each RS encoded symbol is mapped to two 2-tone FSK signals. For a 3-tone FSK modulation, a RS encoding over GF(2^3) may be used. The field elements of a two error correcting (7,3) RS code over GF(2^3) can be generated using a primitive polynomial $p(X) = 1 + X + X^3$. The modulated analog signal samples may be converted to a continuous time waveform using a microcontroller with an integrated Digital-to-Analog Converter and played over a speaker. As an alternative to a hardware interface, for efficient storage and portability of data, the modulated analog signal samples may be converted to an MPEG Layer III (mp3) format using the LAME MP3 encoder [13]. The mp3 format can be decoded and played using compatible speakers.

The transmit/receive chain of the modem is shown in Fig 5. The message to be transmitted is RS encoded. The encoded data is modulated using a 2-tone FSK modulation. The modulated signal samples are passed to an mp3 audio codec for interfacing with a speaker to transmit the modulated analog waveform. The receiver has a front-end bandpass filter whose bandwidth spans the frequencies in the 2-tone FSK modulation for two channels to reduce the out-of-band noise. The spectrum sensor detects the presence of either one of the two channels. The demodulation of the 2-tone FSK signals is performed on a valid channel. A RS decoding on the demodulated data retrieves the original message.

Sensing slot	Beacon broadcast	Encoded Data [Message+Flag + Parity]

Figure 6. Physical layer frame structure

The spectrum sensor needs to be implemented on both the secondary transmitter and receiver so that they are synchronized and are able to communicate at the right time on a common channel. The physical layer frame structure is shown in Fig 6. The Sensing slot is used to sense for available channels. An optional Beacon broadcast time slot can be used to synchronize a transmitter/receiver pair for operation on a common channel. Every channel can be allocated an identifying bit sequence. The Encoded data frame contains the flag sequence $\left[s_1, s_2, s_3, s_4\right]$ which is used to indicate the data encoding and modulation scheme to be used by the SU devices. This can be adapted to the data rate and performance requirements. In general, secondary users may not be aware of the following information about the primary users: a) Operating frequency range and bandwidth and b) Minimum SNR needed to determine the spectrum availability. Hence the main parameters which need to be optimized for the modem are: a) The spectrum sensing time b) The data transmission time and c) Modulation scheme.

B. *Sensing and Access Strategy*

The modem sensing strategy involves sensing for two channels simultaneously. The access strategy was designed for a time-slotted operation, with a periodic sequence of ON/OFF time slots. A physical layer frame is scheduled for transmission for every ON slot. An example channel usage pattern is shown in Fig 7. The gray boxes represent a time-frequency slot occupied by PUs. The white boxes represent a time-frequency slot available for transmission by the SU. The users in the primary network are operated in a synchronous time-slotted manner. The spectrum sensor may sense for the presence/absence of the two channels. If the PU traffic is not slotted, the sensing periodicity needs to be increased so that appropriate time-frequency slots can be identified for data transmission by the SU.

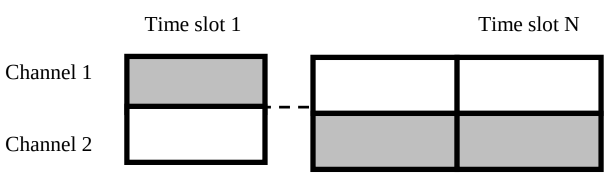

Figure 7. An example slotted channel usage pattern for 2 channels over N time slots.

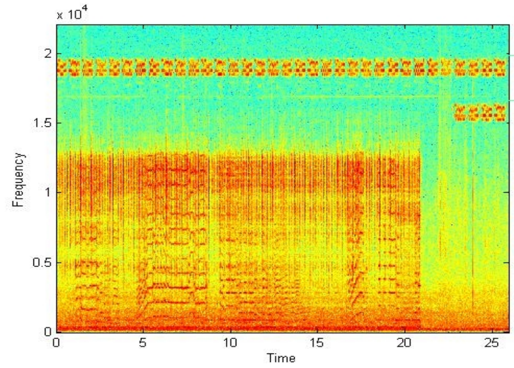

Figure 8. Spectrogram corresponding to UAB transmission over two channels in the presence of interference

Figure 8 shows a spectrogram of the UAB transmission in presence of a TV and Hi-Fi music system as sources of interference. The spectral components near the 20KHz and 16KHz range belong to the 2-tone FSK signal. Channel 1 is close to the 20KHz range and Channel 2 is close to the 16 KHz range. Since there is a strong interference from the audio/music for about 20 seconds, the UAB transmitting modem schedules on only one channel near 20KHz for interference free operation. When the source of interference is turned off, the UAB modem can schedule its transmission on two channels with frequencies closer to 16KHz.

C. *Measurement Results and Performance*

The received signal power at different distances between a speaker with a sensitivity specification of 80dB/W/m and a receiving iPad with a microphone is given in Figure 9. A statistical average of the minimum and the maximum received power at different distances is compared in the plot. The receiving iPad could decode data correctly up to a distance of 30 ft.

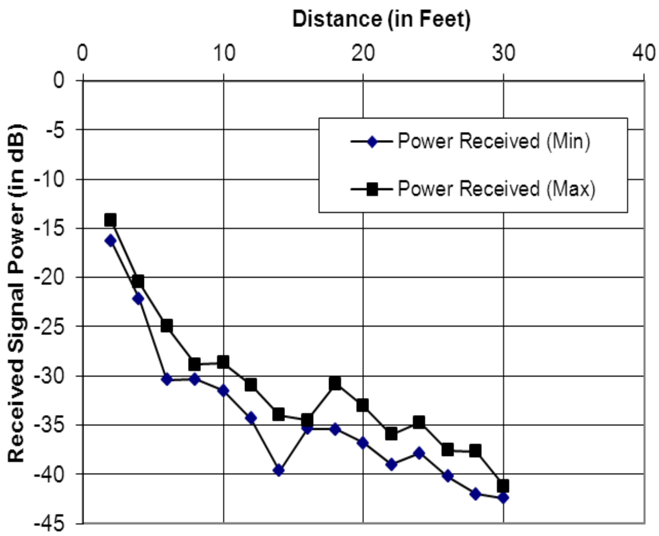

Figure 9. Received signal power at different distances between a speaker and a receiving iPad

A slotted data traffic from the PU was generated on a single channel. The receiving audio modem was tuned to synchronize with the empty time slots to schedule its transmission. The throughput of the receiving modem is defined as the ratio of the total time of sensed empty time slots to the total time of observation. The effective throughput for different sensing durations and a 2-tone FSK signal model is given in Table I.

TABLE I. EFFECTIVE THROUGHPUT FOR DIFFERENT SENSING DURATIONS

	Sensing Duration		
	20ms	13ms	10ms
Distance			
1 feet	0.5297	0.5569	0.5674
2 feet	0.5345	0.5578	0.5679
3 feet	0.5293	0.5544	0.5639
4 feet	0.5360	0.5587	0.5684
5 feet	0.5398	0.5618	0.5709
6 feet	0.5371	0.5601	0.5699

A better throughput is obtained with smaller sensing duration. An improvement can be made on the sensing duration using the method proposed in Section II using the eigenvalues of the covariance matrix of the STFT of the received signal as test statistics for tone detection. This method can improve the throughput for small sensing durations as well as reduce the probability of false alarms/misdetections.

VII. CONCLUSION

A reconfigurable data modem based on the upper audio band has been implemented which can operate on devices with audio interfaces. Examples of system deployment architectures along with the channel usage model are given which can be used for data transmission. Signal models based on multi-tone FSK are proposed, which are optimal for an audio channel and can be easily sensed by an audio spectrum sensor for the presence/absence of valid tones. Spectrum sensing techniques using un-windowed periodograms and eigenvalues of the STFT covariance matrix are proposed. A prototype modem using the periodogram as the tone detector was implemented. The effective throughput of the modem is reported for different sensing durations and distances. The modulation can be changed from a 2-tone to 3-tone FSK for improving the throughput. The low data rate short-range communication modem is implementable in software on mobile devices. Future work involves developing standards defining the system architectures, interfaces and protocols for robust connectivity, service discovery and co-operation for reconfiguration of parameters. The components, which can be standardized include: a) Functional blocks that are necessary for the management and control of radio resources b) Specifications of interfaces between functional blocks and c) Specifications of messages to be exchanged between functional blocks. Applications of the proposed adaptive spectrum aware UAB modem are foreseen in peer-to-peer ad-hoc networks for data exchange in enclosed environments.

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