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An Overview of Range Detection Techniques for Wireless Sensor Networks*

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***Abstract*-Range detection is a key problem in wireless sensor networks (WSN). In this paper, a brief overview is provided for basic techniques for this problem. The fundamental principles for range detection are inherited from RADAR systems, but careful designs need to be made to balance the expected accuracy and the device complexity, considering the limited resources of a WSN. We will introduce some basic measurement principles of range detection for WSN and some available range detection systems. We will also discuss several signal processing algorithms used for range detection.**

***Index Terms* – Range detection; Localization, Wireless sensor network.**

I. Introduction

Target localization is a major application for wireless sensor networks (WSN). Range detection is a key component for localization. In traditional location techniques for RADAR systems, range measurements are complemented with angle measurements for target localization. Large frequency band and expensive equipment, including an antenna array and sophisticated electronic and computing devices, is needed for these systems. In a WSN, however, range measurements are the primary information available for localization. It is not possible to use standard radar techniques because of narrow frequency bands (typically ISM 2.4GHz and 5.8GHz) and limited electronic and computing capability. The fundamental principles for range detection are not much different from those used in radar systems. But new range detection techniques are needed for WSN to balance the resource requirement and measurement accuracy.

In this paper, we introduce some of basic range detection principles that are inherited from earlier RADAR systems (Section II). We then discuss several existing range detection platforms, some experimental and some commercial. We will focus on their setups and approaches, implementations and accuracy results (Section III). Afterwards, we will discuss signal processing algorithms that are commonly used for enhancing measurement accuracy in the presence of multi-paths (Section IV). Some concluding remarks will be reached in Section V.

II. Range Detection Principles

Commonly used range detection techniques employ radio frequency (RF) or acoustic (ultrasonic) signals. Some devices use infrared or global positioning systems (GPS), for example, which we will not discuss in this paper; see [6] for a summary of these techniques. The physical quantities used for range detection are mostly signal travel time and signal power strength. We can divide the detection techniques into several classes according to different measuring principles of these quantities. These techniques include time of arrival (TOA), time difference of arrival (TDOA), round-trip time-of-flight (RTOF) and radio signal strength (RSS).

The RSS technique is based on the propagation decay of a radio signal. The other techniques are based on fact the propagation speed of the signal, either radio or acoustic, is almost constant in the measurement environment, so the range detection problem becomes the propagation time detection problem. The sending time and arrival time of a signal in WSN are usually marked using time-stamping at the MAC layer of the network.

For acoustic signals, the travel time is relatively large, so its direct measurement is relatively easy. For radio signals, the travel time is extremely small, so the travel time needs to be measured indirectly using various modulation techniques. We will introduce linear frequency modulation (LFM) and a variant of it called chirp spread spectrum (CSS) for WSN which is used as an alternative PHY standard in IEEE 802.15.4a.

A. The TOA and RTOF Techniques

The TOA technique is used when the transmitter (target) and the receiver (range sensor) are time synchronized. The setup is shown in Fig.1 (a). The sending time is stamped on the same signal or an auxiliary signal transmitted as the same time. Since the nodes are synchronized, the propagation time of the signal is directly calculated by subtracting the sending time of the signal from the arrival time.

Commonly used time synchronization algorithms for WSN can give an accuracy of microseconds. This accuracy is enough for range detection using acoustic signals. However, for radio signal based range detection, the TOA technique is inadequate because one microsecond error in time

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synchronization will cause about 300m error in distance measurement.

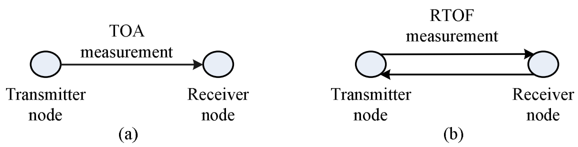

Fig. 1: Range Detection Principles: (a) TOA; (b) RTOF.

When the transmitter and receiver are not time synchronized, the RTOF method shown in Fig. 1(b) can be used. The transmitting node emits a signal and waits for it to reflect back off the target so that the RTOF can be measured for range calculation. Alternatively, the target is also a transmitter/receiver node. When detecting the signal from the first node, the target node acts as a transponder by sending a signal back without any additional processing. The second method makes the reflected signal strong, but the resending creates time delay and is also an error source for distance calculation. This two-way method is similar to the commonly used synchronization algorithm known as Timing-Sync Protocol for Sensor Networks (TPSN).

B. The TDOA Technique

The standard TDOA technique requires a minimum of three nodes. The simplest setup is that three nodes are in a line, as depicted in Fig. 2. Two of them act as receivers and the third one as a transmitter. The transmitter emits one signal and the receivers can then use the arrival differences to calculate the distance between the two receivers.

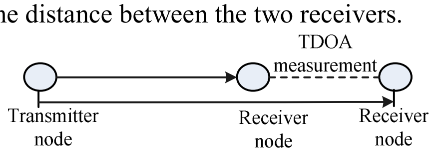

Fig. 2: Range Detection Principle of TDOA.

In a two-dimensional setup, one transmitter and two receivers are used. It is assumed that the receivers are time synchronized with known locations. The TDOA from the transmitter to the two receivers can be used to determine the position of the transmitter.

C. The RSS Technique

RSS is a very simple technique so it has been widely used in WSN. This method simply relies on the fact a radio signal decays as it propagates through medium. No beacon nodes and accurate timing devices are needed and can be used in the presence of certain obstacles because radio signals can pass through many non-metallic objects. A standard radio propagation model is the following path-loss model:

$$PL(d) = 20\log(4\pi d_0/\lambda) + N\cdot 10\log(d/d_0) + X_\sigma$$

where $PL(d)$ is the power loss at the distance of d from the signal source, d is the target distance, d_0 is a reference distance (usually choose to be 1m), λ is the radio wavelength, X_σ is a zero mean Gaussian variable used to characterize measurement noises, and N is the path-loss coefficient. The distance d is calculated once the power loss is measured.

WSN chips such as CC2420 now provide a pin of RSSI (Radio Signal Strength Indicator) and a locating engine based on the path-loss equation. This has helped make the RSS method readily available for WSN. However, the RSS technique suffers from the multipath problem heavily, which is much more serious in the indoor environment. In addition, the path-loss coefficient N varies greatly, depending on the medium, which makes range detection using this method difficult or inaccurate.

D. Linear Frequency Modulation

The linear frequency modulation (LFM) technique transmits a modulated radio signal pulse with its instantaneous frequency changing linearly in time, as shown in Fig.3. This technique is commonly used in frequency-modulated constant-wave (FMCW) radars [5].

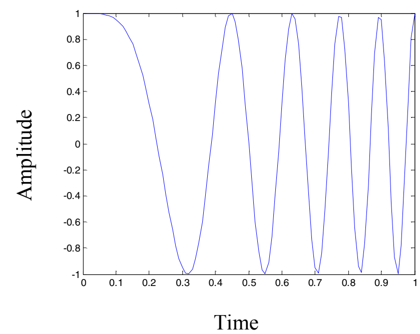

Fig. 3: An LFM Signal

In a FMCW radar, an LFM signal is transmitted and a copy of it is kept as a reference (local) signal. When the transmitted signal arrives at the target and gets reflected back, it will be mixed with the local signal. The frequency difference between the two signals is proportional to the round-trip time, hence is used to compute the target distance. In this way, measuring the frequency differences above will

The Chirp Spread Spectrum (CSS) technique [8], which is an alternative PHY standard for IEEE 802.15.4a, is based on a similar principle. The difference between CSS and FMCW is that CSS uses an improved version of round-trip time measurement scheme called Symmetric Double Sided Two Way Ranging (SDS-TWR) [8].

The principle of SDS-TWR is shown in Fig. 4.

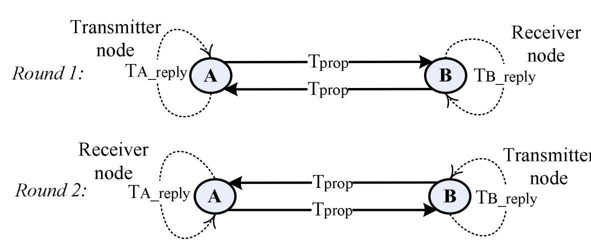

Fig. 4: The SDS-TWR Principle

The two nodes can be used for both transmitting and receiving and they are not necessarily time synchronized. Two round-trip transmissions are done. The two nodes have their roles reversed in the two rounds. In each round, the total round-trip time includes the round-trip propagation time and the processing times at both ends. The processing time of each node is about hundreds of microseconds whereas the propagation time is only about tens of nanoseconds.

The propagation process can be written as

$$T_{prop} + T_{B\_reply} + T_{prop} = T_{A\_wait}$$
$$T_{prop} + T_{A\_reply} + T_{prop} = T_{B\_wait},$$
$$T_{prop} = \frac{1}{4}\left(T_{A\_wait} + T_{B\_wait} - T_{A\_reply} - T_{B\_reply}\right).$$

Here, T_{prop} is the signal propagation time, $T_{\bullet\_reply}$ is the processing time of the node, $T_{\bullet\_wait}$ are the time between sending out a signal and receiving a signal back. Among those time measurements, $T_{\bullet\_reply}$ and $T_{\bullet\_wait}$ can be measured by their own local oscillator. But the measurement is biased caused by oscillator's offset $\delta T_{(\bullet)}$. The reason for using two rounds of transmission is to remove the offset.

III. Range Detection Platforms

A. Cricket indoor location system

The Cricket system [1] contains a beacon node, which is fixed on the ceiling with a known position, and listening nodes, whose locations need to be determined. Two signals are transmitted, a radio signal and an ultrasonic one. The distance between the beacon and a listening node is measured using the TDOA technique. The beacon node transmits the radio and ultrasonic signals concurrently and the listening node will record the different arrival times of two signals caused by their different propagation speeds. The time difference of arrival is then used to compute the distance.

The ranging accuracy of a Cricket system is about 5 cm and the locating accuracy is about 10 cm. The high accuracy is due to the use of an ultrasonic signal. But the effective ranging distance of ultrasonic signals is short, only about 10 m at normal temperature. This system also suffers in a non-line-of-sight (NLOS) environment because of the poor penetrating ability of ultrasonic signals.

B. Acoustic Embedded Networked Sensing Box (ENSBox)

ENSBox [2] is developed to address the problem of acoustic source localization. Every node of ENSBox is a beacon node and they realize a self-localization scheme before set up for locating the target source.

Its range detection scheme of the self-localization process is based on the TOA measurement of a chirp signal arrived at different nodes. The transmitter node plays an acoustic chirp signal through a microphone and sends out the starting time in a radio signal via a chirp notification packet. The packet is delivered through the whole network by flooding services. When each receiver node "hears" the chirp signal, they will read the start time of the chirp from the notification packet and calculates the time-of-flight of the chirp signal.

The ENSBox uses a conversion approach rather than synchronization scheme between transmitter and receiver nodes. The objective of a traditional time synchronization scheme is to control the oscillator rate of each node to be the same. The conversion approach, however, allows an independent clock for each node and "translates" the event time to a local node in each hop by a specialized protocol.

The ENSBox is claimed to have a ranging accuracy of 5cm in 80m by 50m area outdoor. It was also tested in a forest environment achieving a 3D localization accuracy of 33cm.

There are also some other platforms used for audio source localization, e.g., counter-sniper system [3]. Ref. [2] compared the performance of ENSBox and Counter-sniper in detail. The localization systems based on acoustic signals all suffer from the problem of limited bandwidth and interference from other acoustic sources in the same environment.

C. Radio Interferometric Positioning system (RIPs)

The Radio Interferometric Positioning system (RIPs) reported in [4] uses the measurement of the received phase difference for detecting the range between each pair of nodes. The original principle of interferometric measurement is that the phase differences received by two nodes from the same emitter are caused by the different distances from emitter to each node. In other words, the phase differences contain information about the distance.

But the traditional methods need a complex device, such as directional antenna, and tight time-synchronization for each node, which are barely possible to realize on a WSN. The RIPs tries to realize intereferometric in commercial WSN equipment based on two improvements.

Firstly, it measures the relative phase differences of two nodes from two emitters rather than only one emitter as in traditional methods. The phase differences can be depicted as a function of the four distances between each transmitter-receiver pair. As shown in Fig. 5, the phase differences between the two receiver nodes C and D can be written into such an equation as:

$$phase\_difference = 2\pi \frac{d_{AD} - d_{BD} + d_{BC} - d_{AC}}{\lambda}.$$

Here, λ is the wavelength of the carrier.

Phase estimation can be obtained by commercial algorithms. Then the relative distance can be calculated by measuring some more groups of four nodes. It has been proved that at most $\frac{3}{2}(n-2)(n-3)$ independent interference measurements will be enough for calculating the unknown distance measurements in a network with n nodes.

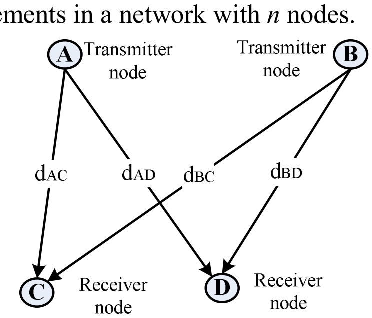

Fig. 5 Ranging principle of RIPs

Secondly, the two emitters transmit two nearby frequencies. Then multiplying these two frequency signals at receiver will get an interferometric low-frequency envelope, which could be measured from the RSSI (Radio Signal

Strength Indicator) pin of the commercial chip such as CC1100. It is worth noting that the RSSI here is just used for measuring the received interferometric signal envelope rather than ranging based on the model of RSS.

The average accuracy of the system is announced to be 3 cm and the largest error is about 6 cm. But as Ref. [2] points out, the RIPs might be sensitive to the multipath indoor and the test results are perhaps taken in the outdoor.

D. Local Positioning Radar (LPR)

Local Positioning Radar (LPR) and an improvement version LPR-B [5] worked like a FMCW radar, the principle of which has been discussed in part D of Section II, at the ISM 5.8GHz band. The receiver in classical FMCW radar just acts as a transponder that relays the signal back without any special processing. But the receiver in LPR is active since it can be self-synchronized with the transmitter.

The out-of-synchronization between nodes is caused by the rate drift of oscillator. The chirp signal produced by the local oscillator has both time and frequency drift with the received signal. To offset the drift in LPR, the transmitter sends an up-sweep chirp signal followed by a down-sweep chirp signal to the receiver. The local chirp signal mixed with the received up- and down-sweep signals respectively and the output of each mixer is a mid-frequency signal. The time and frequency drifts are contained in the output signal and can be revealed from the mid-frequency. Estimating the output mid-frequency by algorithms such as FFT (Fast Fourier Transform), the drifts can be offset.

The LPR system can potentially achieve the standard deviation well below 1cm and the average error is about 3cm in the range of 5 to 25m.

The challenge for using LPR in an indoor environment is the multipath problem. Although a multipath mitigation scheme is introduced in [5], its performance is based on a wide bandwidth.

E. RESOLUTION Project

RESOLUTION [6] is a sponsored project in order to get a centimeter level positioning system in an area up to 1000㎡. The measurement they used is also RTOF and their scheme is similar to FMCW radar.

It is interesting that their receiver acts differently from both FMCW and LPR. Once the receiver detects a transmitted signal, it will drop its own local frequency and regard the received signal as local oscillator frequency compared to the relay and pre-synchronization operation in FMCW and LPR respectively.

The system named HPLS [6] uses 150MHz bandwidth around 5.8GHz. It can achieve the standard deviation of 4cm in an ideal anechoic chamber and 18cm in a 800 ㎡ conference hall.

The HPLS system also suffers from the problem of multipath in an indoor environment. Also, the system consists of a lot of specialized IC designs [7], which is difficult to extend to other platforms.

IV. Range Detection Algorithms

We have introduced the common used measurement principle and several available platforms based on these principles above. Actually, no matter what kind of measurement principle is used, the ranging system all faces a problem that the received signal is interfered by multi-emitter and/or multipath propagations.

What we demand from the received interfered signal is the line-of-sight (LOS) path component since only the arrival time of the LOS path can be used for calculating the direct distance. The range detection problem turns out to be a problem of signal propagation time delay estimation.

Traditionally, the estimation of propagation time delay is obtained by cross-correlation method, e.g. a matched filter followed by an envelope detector. However, when the operating bandwidth is limited and the correlation function is not narrow enough, the width of main lobe of the correlation might be larger than the difference of time delay between LOS path and other path. Those paths cannot be distinguished then and the range detection effect will be disappointing.

Considering the propagation channel impulse response (CIR) like

$$h(t)=\sum_{k=0}^{m-1}\alpha_k\delta(t-\tau_k),$$

where m is the number of the multipath, α_k is a attenuation coefficient, τ_k is the time delay and τ_0 is the time delay of LOS demanded path, i.e., TOA. Taking Fourier transform of the above CIR obtain

$$H(f)=\sum_{k=0}^{m-1}\alpha_k e^{-j2\pi f\tau_k}.$$

As [10] pointed out, if we exchange the role of time and frequency of this equation, it will be changed into a harmonic signal model

$$H(\tau)=\sum_{k=0}^{m-1}\alpha_k e^{-j2\pi f_k\tau}$$

which is well known in spectral estimation area. Spectral estimation algorithms such as MUSIC can be used for the time delay estimation of the frequency channel response.

A. Multiple Signal Classification (MUSIC)

The MUSIC algorithm was first introduced by R. Schmidt in a workshop of American Army in 1979 and the original document was published in 1986 [9]. MUSIC is an abbreviation of MUltiple SIgnal Classification.

In [9], the MUSIC algorithm was originally used for detecting different arrival angles on an antenna array. But it can be extended to estimate other constant parameters, such as the propagation time-delay, by using a “snapshot” of the output.

The common used methods before MUSIC were Maximum-Likelihood (ML) and Maximum-Entropy (ME). But they both suffer from a big problem that they use an incorrect model, such as AR, of the measurements.

Considering the signal model,

$$X(n) = H(n) + W(n) = \sum_{k=0}^{p-1} \alpha_k e^{-j2\pi f \tau_k} + W(n)$$
$$= VA + W$$

where $X(n)$ is the sampled data from the channel impulse response (CIR) with Fourier transform $X(f)$. Here,

$$A = [\alpha_0 e^{-j2\pi f \tau_0} \ \alpha_1 e^{-j2\pi f \tau_1} \ \cdots \ \alpha_{p-1} e^{-j2\pi f \tau_{p-1}}]^T$$
$$V = [v(\tau_0) \ v(\tau_1) \cdots v(\tau_{p-1})]$$
$$v(\tau_k) = [1 \ e^{-j2\pi f \tau_k} \cdots e^{-j2\pi(M-1) f \tau_k}]^T$$

M is the number of samples. MUSIC algorithm is based on the decomposition of the covariance matrix of the sampled data,

$$E\left\{ x x^H \right\} = VAA^H V^H + \sigma^2 I \, .$$

It can be shown that the smallest eigenvalue of the sample covariance matrix $E\left\{ x x^H \right\}$ repeat $M - p$ times and is equal to σ^2. The eigenvectors, say $e(i)$, corresponding to this eigenvalue span a noise subspace while the subspace spanned by eigenvectors corresponding to other eigenvaules forms a signal subspace.

Considering the fact that $e(i)$ is orthogonal to signal space and the vector $v(\tau_k)$ containing time delay lies in the signal space, we have $e(i)^H v(\tau_k) = 0$. Defined a function in the form of inverse of the Euclidean distance to the noise space spanned by $e(i)$, the point where the function towards infinity large would correspond to the $v(\tau_k)$.

Ref. [10] compared the MUSIC algorithm with commercial TOA estimation technique, channel impulse response (CIR) estimation and cross-correlation techniques with DSSS (Direct-Sequence Spread-Spectrum) signals, based on experiments. Nearly 80% of ranging measurement errors is below 3m when 80MHz bandwidth is used by MUSIC. This is 5% and 20% better than above methods respectively.

The MUSIC algorithm is a kind of model-based method and is sensitive to the differences between the actual received signals and the model. But compared to the common used Max-Likelihood and Max-Entropy algorithm, the MUSIC still shows a great improvement in resolution.

B. Minimum-norm method

The minimum-norm method [16] can also be used for the time delay signal model depicted earlier in this section. The traditional linear prediction (LP) method was used for fitting multi-sinusoid signal models to observed data. But the LP based estimation method is greatly affected by noises. The minimum-norm method improves the estimation performance by singular value decomposition (SVD) of the LP data matrix.

Considering the same signal model of MUSIC from view of the geometry, the minimum-norm method searches for the whole parameter space spanned by $a(\theta)$ to minimize $\left| a^H(\theta) w \right|^2$, where w is a linear combination of vector of noise space as $w = \sum_{i=1}^{N-P} c_i u_i$. The value of the vector $\{c_i\}$ is the solution of LP that minimizes the Euclidean norm $w^H w$.

The common scheme of using minimum-norm method for estimating the time delay is reported as [11]. The time delay is estimated from the transfer function of the propagation medium. The transfer function is sampled N times and the minimum-norm method is used to find a prediction vector of the sampled sequence. When the prediction vector is obtained, the time delay can be easily calculated from the transfer function.

The minimum-norm method performances well even at low SNR cases. However, the step of channel transfer function estimation affects the estimation accuracy of the method in a certain extent.

C. Matrix Pencil (MP) algorithm

The Matrix Pencil (MP) algorithm [13] is also a matrix prediction approach like minimum-norm method. But it deals directly with the sampled signals rather than signal covariance matrix.

The principle of MP algorithm is to find a rotation matrix that can reduce the rank of the defined pencil matrix by one. The interesting feature of the rotation matrix is its diagonal elements containing the time delay we want.

The specific process of the algorithms is as below. Firstly, sampled N times from the $X(f)$ obtaining a sequences in frequency area,

$$y_f = \left[X(f) \ \ \cdots X(f+m-1) \right]^T .$$

Here, $M = N - m + 1$ and $f = 1, \ \cdots, N$.

Form the pencils of vectors like $p_f = y_f - \lambda y_{f-1}$. The matrix formed by $\left\{ p_f \right\}$ called pencil matrix is full rank except when $\lambda y_{f-1} = y_f$. It can be easily improved that the λ that reduces the rank of $\left\{ p_f \right\}$ by one equivalent to $e^{-j2\pi f \tau_i}$. The diagonal matrix $\Phi = diag\left\{ e^{-j2\pi f \tau_i} \right\}$ is called a rotation matrix. The time delay is contained in the phase shift of the diagonal element $e^{-j2\pi f \tau_i}$, hence the problem of estimating time delay turns out to be a problem of finding the generalized eigenvalue λ of the pencil matrix $\left\{ p_f \right\}$.

There are several techniques for finding λ [13]. The principle of all these techniques is retrieving the diagonal matrix $\Phi = diag\left\{ e^{-j2\pi f \tau_i} \right\}$ which forms Y_1 and Y_2 as $Y_1 = E_1 A \Phi E_2$ and $Y_2 = E_1 A E_2$. Here, E_1 and E_2 are full rank matrices, and Y_1 and Y_2 defined as

$$Y_1 = \left[y_M, y_{M-1}, \cdots, y_1 \right]$$
$$Y_2 = \left[y_{M-1}, y_{M-2}, \cdots, y_0 \right].$$

Ref. [13] points out that one of those techniques is the famous ESPRIT algorithm [15] which can also be recognized as a variation of the matrix pencil algorithm. ESPRIT turns the problem to find p eigenvectors corresponding to p zero eigenvalues of a Hermitian matrix, which is formed by two $(N-M)\times p$ sub-matrixes of the $(N-M)\times M$ matrixes Y_1 and Y_2 respectively, and p is the number of the paths in the received signal.

The MP algorithm requires much less computation load compared with the MUSIC algorithm. Another advantage of MP is its ability to calculate the correlated components while MUSIC requires the components in the subspace to be uncorrelated.

The MP algorithm is utilized for TOA estimation in CSS system [14]. They chose the channel model 1 of IEEE 802.15.4a and used MP for TOA estimation based on the minimum mean square error (MMSE) estimation of CIR. The simulation results showed that the MMSE-MP scheme outperforms the MUSIC and can get an average ranging accuracy below 70cm within the coverage of channel model 1 from a 7m to 20m residential line-of-sight area.

V. Conclusion

The ranging techniques in WSN are inherited from the popular ranging schemes in traditional communication systems such as FMCW and RSS for radar and cell-phone locating systems. But the ranging problem of WSN still has its own characters, e.g. TDOA of radio and ultrasonic signals used by Cricket system, subjected to the limitation of its small size and low power but also benefited from its large number and wide-area deployment.

The signal processing algorithms used for detecting LOS path from the received signal also needs to consider the trade-off between the accuracy requirement and the limited WSN resources. We have introduced several popular algorithms used for improving the TOA estimation of the received signals. Actually, improving the resolution of other measurement schemes, such as TDOA, RTOF and FMCW, is identical to that for TOA in nature.

Though these algorithms have brought improvement to the range detection, there are still several problems unsolved: Firstly, the ranging still doesn't satisfy the need of accuracy in some complex environments. Most platforms and schemes can reach the accuracy under 1m, but this result will be much worse in a complex indoor environment subject to the multipath problem. Secondly, some improvements of ranging techniques are depending on the big bandwidth while there is only 80MHz bandwidth at the ISM 2.4GHz band for WSN. Thirdly, exchanging messages too often will create a heavy load for the network. Take RTOF for example, the ranging data need a two-way propagation which means a double load for the network. And the SDS-TWR used by CSS is even worse since it needs four-way propagation for ranging. This extra load must be considered carefully for WSN since the capacity and the immediate dada rate of WSN is very limited.

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