Electromagnetic propagation logging while drilling data acquisition method based on undersampling technology

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Received 24 April 2021, revised 26 June 2021
Accepted for publication 10 August 2021

Abstract
With the development of complex and unconventional reservoirs, oil and gas exploration becomes increasingly difficult. Highly deviated wells/horizontal wells are widely used. The electromagnetic propagation logging while drilling (LWD) is more effective in complex geological environment detection owing to geological orientation and real-time formation evaluation. However, its operating frequency is generally at the MHz level. Traditional acquisition techniques require an analogue to digital converter with high sampling rates, which will introduce complex circuit structures and increase sampling costs. The undersampling technology has overcome these disadvantages. The difficulties in the undersampling technology include the selection of an undersampling frequency and the acquisition of a signal correction coefficient. The range of undersampling frequencies and a correction coefficient has been developed to process the electromagnetic propagation LWD measurements in this paper. The range of undersampling frequency ensures the validity of the sampled data. The correction coefficient ensures that different frequency signals use the same undersampling frequency to obtain the same frequency recovery signal. The correctness of these parameters is verified by simulation and field data examples. The range of undersampling frequency and a correction coefficient has been applied, improving the data stability and providing reliable technical support for the exploration and development of unconventional oil and gas.

Keywords: electromagnetic propagation LWD, sampling technology, undersampling, digital phase sensitive detection, sampling frequency

1. Introduction
During drilling, the electromagnetic propagation logging while drilling (LWD) tool is used to measure the electrical parameters of formation. At this time, as the formation has not been completely invaded by the drilling fluid, the resistivity information obtained can reflect correct electrical characteristics of stratum (Wang et al. 2007), and combine deep and shallow multi-scale geological and geophysical information to analyse the deep structures of the stratum (Di et al. 2021). Teleco officially launched LWD tools in
1978, marking the commercialisation of LWD technology (Spinnler & Stone 1978). With the introduction of phase shift and attenuation resistivity concepts (Clark et al. 1988), the calculating theory of attenuation and phase shift to reflect formation resistivity has gradually matured. The electromagnetic propagation measuring tool with multi-coil spacing and multifrequency (Rodney et al. 1991) and new azimuthal electromagnetic propagation resistivity tool (Wang et al. 2006) have been introduced successively. The electromagnetic propagation LWD technology has important practical value in directional drilling and formation boundary detection (Li et al. 2005; Bell et al. 2006; Meyer et al. 2008), and has gradually become an important part of LWD. At present, Schlumberger, Halliburton and Baker Hughes have launched the PeriScope, ADR and AzitTrak instruments, respectively. (Omeragic et al. 2005; Bittar et al. 2009).

The sampling technology is one of the most important links in the process of acquiring electromagnetic propagation LWD data. For high-frequency signal during logging (for example, 400 kHz or 2 MHz), the traditional method obtains the amplitude and phase information of the received signals by combining the superheterodyne technology with the undersampling technology (Rodney & Wisler 1986; Zhang & Pang 2014), and has several disadvantages such as complex design circuit, circuit noise and long processing time (Wu 1996; Peng 2017). Undersampling technology is proposed for the processing of electromagnetic propagation LWD data in this study. It has several advantages, such as simpler circuit implementation and effective reduction of the noise generated by a circuit.

A freely selected undersampling frequency cannot recover the amplitude and phase information of the received signal. The determination of the undersampling frequency is critical for the application of the undersampling technology. In this study, the selection range of the undersampling frequency is determined through equation derivation and in combination with the principle of digital phase sensitive detection (DPSD). To obtain the recovered signals at the same frequency, the received signals at different frequencies must be undersampled. After analysing the physical meaning of the undersampling technology, the signal correction factor is proposed. The corrected signals at different frequencies are sampled at the same undersampling frequency to obtain the recovered signals with the same frequency. To simplify the calculation and improve efficiency, the data are stored and calculated in a matrix. Finally, the processing effect of both the traditional sampling technology and undersampling technology are obtained through comparison simulated or measured data, respectively. The processing result of the undersampling technology is more stable in engineering application. With the application of this technology, the reliability output of the electromagnetic propagation LWD tool will be improved. The simulated and measured results provide theoretical support for the application of the undersampling technology.

2. Principle of data acquisition and signal correction

The electromagnetic propagation LWD tool indicates the formation resistivity by the attenuation and phase shift of received signals. The conventional superheterodyne technology obtains the amplitude and phase information of the received signal by frequency reduction. The circuit designed with superheterodyne is featured by complex structure and ease of introducing circuit noise interference. The undersampling technology can implement a simple circuit design, which overcome the shortcomings of the superheterodyne technology and obtain stable results.

2.1. Superheterodyne principle

The traditional data processing method based on the superheterodyne technology is shown in the following. First, the received high-frequency signals are mixed with the oscillation signals generated by the local oscillator. Second, the mixed signal is then transmitted through a band-pass filter to obtain a low-frequency signal. Finally, the low-frequency signals obtained are subjected to the DPSD to obtain the amplitude and phase information of the received signals.

The principle of the superheterodyne technology can be explained by equation calculation in detail. The received and local oscillation signals are set as follows:

\[ x(t) = KU_0 \cos(2\pi F_0 t + \theta), \]

\[ l(t) = \cos(2\pi F_t t), \]

where, \( K, U_0, F_0, F_t \) and \( \theta \) refer to magnification times, amplitude of received signals, measuring time, natural frequency of the received signals, frequency of the oscillation signals, phase of received signals.

The received signals are mixed with the oscillation signals, and the high-frequency signals are filtered out by the band-pass filter to obtain the equation:

\[ x_{\text{output}}(t) = x(t)l(t) \rightarrow B \text{ and-Pass Filter} \]

\[ \frac{KU_0}{2} \cos(2\pi(F_t - F_0)t + \theta). \]

We see from equation (3) that the signals processed by the superheterodyne technology are reduced to low-frequency signals through a band-pass filter, and then low-frequency signals are oversampled and subjected to the DPSD to obtain the amplitude and phase information of the received signals. The data processing based on superheterodyne involves cumbersome steps and complex implementation circuit.
Through investigation and analysis, data processing based on the undersampling technology can effectively overcome the shortcomings of the superheterodyne technology and obtain the amplitude and phase information of the received signals.

### 2.2. Principle of undersampling technology and undersampling frequency determination

Data processing based on the undersampling technology is shown next. First, signals are sampled. Then they are subjected to DPSD to extract the amplitude and phase information of signals. Accurate received signals can be obtained at a low sampling cost. Determining the undersampling frequency is critical to the application of the undersampling technology. An effective sampling frequency provides a guarantee for accurately extracting amplitude and phase information.

The received signals after undersampling are obtained as follows:

\[
x(n) = KU_0 \cos \left( 2\pi \frac{F_0}{F_s} n + \theta \right), \tag{4}
\]

where \( F_s \) refers to the undersampling frequency and \( n \) refers to the sampling points. The selection range of \( F_s \) is determined using the DPSD process. The receiving time of the receiving coil is \( t \) (generally several hundred milliseconds), and the number of sampling points is obtained from the equation \( M = F_s \times T \). The reference signals are expressed as follows:

\[
r_1(n) = U_r \cos \left( 2\pi \frac{F_0}{F_s} n \right), \tag{5}
\]

\[
r_2(n) = U \sin \left( 2\pi \frac{F_0}{F_s} n \right). \tag{6}
\]

The amplitude of the received signals is calculated based on the principle of DPSD, as shown next:

\[
U_{01} = \frac{1}{M} \sum_{n=1}^{M} x(n) \times r_1(n), \tag{7}
\]

\[
U_{02} = \frac{1}{M} \sum_{n=1}^{M} x(n) \times r_2(n). \tag{8}
\]

First, calculating \( U_{01} \), substituting equations (4) and (5) into (7), and converting the sine and cosine functions into exponential functions, we obtain:

\[
U_{01} = KU_0 U_r \frac{1}{F_s T} \left\{ \frac{\cos(\theta)}{4} \sum_{n=1}^{F_s T} \left( e^{i4\pi \frac{F_0}{F_s} n} + 2 + e^{-i4\pi \frac{F_0}{F_s} n} \right) \right\}, \tag{9}
\]

Let’s discuss equation (9) under different situations: when the equation \( \frac{F_0}{F_s} = k/2 \) \((k > 1, k \in Z)\), the ratio of geometric progression in equation (9) is 1, and we obtain:

\[
U_{01} = KU_0 U_r \cos(\theta). \tag{10}
\]

When the equation \( \frac{F_0}{F_s} \neq k/2 \) \((k > 1, k \in Z)\), we obtain:

\[
U_{01} = KU_0 U_r \frac{1}{F_s T} \left\{ \frac{\cos(\theta)}{4} \sum_{n=1}^{F_s T} \left( e^{i4\pi \frac{F_0}{F_s} n} \right) \right\}. \tag{11}
\]

When the frequency of the logging signal is calculated by \( F_0 = 2 \text{ MHz} \) or \( F_0 = 400 \text{ kHz} \), \( F_s T \in Z \), and equation (11) is simplified, we obtain:

\[
U_{01} = \frac{1}{2} KU_0 U_r \cos(\theta). \tag{12}
\]

In summary,

\[
U_{01} = \begin{cases} 
KU_0 U_r \cos(\theta) & \left( \frac{F_0}{F_s} = \frac{k}{2}, k > 1, k \in Z \right)
\end{cases}.
\tag{13}
\]

Calculating \( U_{02} \) similarly, we obtain:

\[
U_{02} = \begin{cases} 
0 & \left( \frac{F_0}{F_s} = \frac{k}{2}, k > 1, k \in Z \right)
\end{cases}.
\tag{14}
\]

According to equations (13) and (14) we obtain:

\[
U_s = \begin{cases} 
\text{no result} & \left( \frac{F_0}{F_s} = \frac{k}{2}, k > 1, k \in Z \right)
\end{cases},
\tag{15}
\]

\[
\theta = \begin{cases} 
\text{no result} & \left( \frac{F_0}{F_s} = \frac{k}{2}, k > 1, k \in Z \right)
\end{cases}.
\tag{16}
\]

From equations (15) and (16), we know that the undersampling frequency is related to the frequency of received signals, and should satisfy equation (17):

\[
\frac{F_0}{F_s} \neq \frac{k}{2} \left( k > 1, k \in Z \right). \tag{17}
\]

Equation (17) gives the selection range of the undersampling frequency to ensure accurate extraction the amplitude and phase information. The superposition calculation
can eliminate the noise unrelated to the reference signals. The simulation results show that the recovery effect can be achieved under noise conditions.

Equation (17) provides the selection range of the undersampling frequency. Combined with the characteristics of multifrequency transmission of electromagnetic propagation LWD instrument. To obtain the same frequency recovery signal, different undersampling frequencies should be set for different frequency transmission signals according to equation (17), which increases the difficulty of firmware writing. In the next section, the correction coefficient is proposed through analysis, such that signals at different frequencies obtain the recovered signals with the same frequency and use the same undersampling frequency.

2.3. Correction coefficient of received signal

In this section, the undersampling frequency of the received signal 1 ($F_{01} = 400$ kHz) is determined according to the number of sampling points in a single cycle of the recovered signal. The received signal 2 ($F_{02} = 2$ MHz) is collected by the above undersampling frequency. However, the sampling signal does not satisfy the number of sampling points in a single cycle of the recovered signal. Therefore, the correction coefficient is determined by analysis.

When the conditions in equation (17) are met, the frequency of the received signals and the undersampling frequency can be expressed by $p \times F_0 = q \times F_0'$, where $p$ and $q$ are positive integers and $q > p$. The lowest spectral component of the undersampling result meets the equation $F_s = F_0 - n \times F_0'$, where $F_s$ is the frequency of the received signals and $n$ is rounded to the nearest integer of $q/p$. Thus, $q/p = n + a/p(|a| < p/2)$. Therefore, we obtain:

$$F_s = |F_0 - n \times F_0'| = \frac{|a|}{p} \times F_0'. \quad (18)$$

The physical meaning of undersampling can be taken from equation (18), and the number of sampling points of recovered signals in the $|a|$ cycle is $p$. After undersampling, the frequency of the recovered signals decreases. The amplitude and phase information of the received signals are still preserved. The relationship among frequency of the undersampled, frequency of recovered and received signals is as follows:

$$F_s = \left(\frac{p}{np + a}\right) \times F_0' \quad (19)$$

$$F_x = \left(\frac{|a|}{np + a}\right) \times F_0. \quad (20)$$

To simplify the calculation of DPSD and firmware writing, we consider $a = 1$. The same sampling frequency $F_s$ and sampling points $p$ are applied for different signal frequencies ($F_{01} = 400$ kHz, $F_{02} = 2$ MHz) in the logging process. These conditions are difficult to meet. Through analysis and derivation, we provide a processing method where the same sampling frequency $F_s$ and sampling point $p$ are applied for signals at different frequencies.

Under the conditions $a = 1$ and $p = 8$, we select sampling frequency $F_s$ that the low-frequency signals meet eight sampling points per unit cycle. According to equation (18), we can obtain $F_{01}/F_0' = q_1/8 = n + 1/8$, $F_{02}/F_0' = q_2/8 = 5 \times q_2/8 = 5n + 1 - 3/8$ and the sampling frequency $F = 8F_{01}/(8n + 1)$. The frequency is different after undersampling, which can be expressed as follows:

$$x_1(k) = KU_0 \cos \left(\frac{1}{4} k \pi + \theta\right), \quad (21)$$

$$x_2(k) = KU_0 \cos \left(-\frac{3}{4} k \pi + \theta\right). \quad (22)$$

Equations (21) and (22) are the results of low- and high-frequency signals. As mentioned above, only low-frequency signals meet the given requirements. The high-frequency signals can meet the given requirements by analysing the given correction method. Equation (22) can be written as:

$$x_2(k) = (-1)^k KU_0 \cos \left(\frac{1}{4} k \pi + \theta\right). \quad (23)$$

Comparing equations (23) and (21), when $k$ is considered an odd number we can obtain the equation $x_2 = -x_1$. When $k$ is considered an even number, we obtain the equation $x_2 = x_1$. A correction coefficient $A$ is then introduced for collecting the high-frequency signals, and the corrected high-frequency signals meet the requirement that the number of sampling points per cycle is 8.

$$A = \begin{cases} x(k) = -x(k) & k \text{ odd} \\ x(k) = x(k) & k \text{ even} \end{cases} \quad (24)$$

The introduction of the correction coefficient ensures that the same undersampling frequency is applied for high- and low-frequency signals to obtain sampling results with the same frequency. The correction coefficient simplifies the firmware writing and fixes the reference signals during DPSD. Matrix storage can also simplify the subsequent calculation when the number of sampling points per cycle is known.

As the number of sampling points per unit cycle is known, the process of simplifying the calculation through matrix storage is shown next. There are $m$ cycles for undersampled signals. Data from each cycle are added to the sampling position in the first cycle sequentially, and we obtain:

$$X = \frac{1}{F_s T} \begin{pmatrix} x(1) + x(p + 1) + \cdots + x((m - 1)p + 1) \\ x(2) + x(p + 2) + \cdots + x((m - 1)p + 2) \\ \vdots \\ x(p) + x(2p) + \cdots + x(mp) \end{pmatrix}^T. \quad (25)$$

According to the number of sampling points, we set the reference signal matrix as follows:
We can obtain the equations $U_{01} = XR_1$ and $U_{02} = XR_2$. The amplitude and phase information of the received signals can be calculated based on the DPSD principle. The correction coefficient and matrix storage can simplify the application of the undersampling technology, reduce the complexity of data processing and provide theoretical support for instrument application.

### 3. Simulation results

To verify the correctness of the equations shown and the effectiveness of the undersampling technology. The simulation is divided into two parts: verification of the correctness of equation (17) and testing the processing effect of the undersampling technology. In the first part of the simulation, attention is paid to the recovery of signal amplitude and phase information at different undersampling frequencies. Noise is not introduced during simulation. In the second part of simulation, the effects of the undersampling processing and traditional processing are compared. In the second part, the Gaussian white noise is introduced to the received signals during simulation.

#### 3.1. Undersampling frequency verification

The setting of related parameters is shown in table 1. $F_s = 160$ kHz and $F_s = 110$ kHz, respectively, represent two cases: unsatisfying and satisfying equation (17). When no noise is introduced, we ensure that there is a cycle signal after undersampling. Here, we set the receiving time of signals to 1 ms. When $F_s = 160$ kHz, the received and undersampled signals are shown in figure 1. The comparison between processing results and original data is shown in figure 2.

When the equation $F_s = 110$ kHz, the original signals and undersampled signals are shown in figure 3. The comparison between recovered signals and original signals is shown in figure 4.

It can be seen from the processing result diagrams of different undersampling frequencies that the equation $F_0 / F_s = k / 2(k \geq 1, k \in Z)$ is true. This indicates that it can accurately recover the amplitude and phase information of the received signals. To show the results more clearly and precisely, the data related to the simulation process are presented in table 2, where ‘Y’ denotes recovery and ‘N’ no recovery.

| Data type                        | Value          |
|----------------------------------|----------------|
| frequency of received signal     | 2 MHz          |
| amplitude of received signal     | 1 V            |
| phase of received signal         | 30°            |
| frequency of undersampling       | 110 kHz, 160 kHz|

Table 1. Related parameters in the simulation process

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in electromagnetic propagation LWD. The attenuation and phase shift obtained through the undersampling technology are compared with the calculation results obtained through the traditional methods to verify the calculation effect of the undersampling technology. The attenuation and phase shift are defined as:

\[
\text{ATT} = 20 \log_{10}\left(\frac{|V_{Rx1}|}{|V_{Rx2}|}\right),
\]

\[
\text{PS} = \text{arg}(V_{Rx2}) - \text{arg}(V_{Rx1}).
\]

Here, ATT represents attenuation and PS represents phase shift. \(V_{Rx1}\) represents the received signals from the near receiving antenna and \(V_{Rx2}\) represents the received signals from the far receiving antenna.

### 3.2. Analysis of undersampling effect

According to the introduction in section 2.3, the number of sampling points per cycle for the recovered signal is set at eight. The undersampling frequency is selected as 128 kHz through analysis. The attenuation and phase shift calculated through the undersampling technology are compared with those calculated using traditional methods. The setting of related parameters is shown in table 3.

The two signals represent the near receiving coil and far receiving coil (marked as Rx1 and Rx2, respectively). The amplitude and phase of the received signals for the near receiving coil are 1 V and 30°, respectively. The amplitude and phase of the received signals for the far receiving coil are 0.5 V and 45°, respectively. The received, undersampled and recovered signals after processing are shown in figure 5.
Table 2. Comparison of different sampling frequency processing result

| Frequency (kHz) | Initial amplitude (V) | Initial phase (°) | Recovery amplitude (V) | Recovery phase (°) | Amplitude recovery effect | Phase recovery effect |
|----------------|-----------------------|------------------|------------------------|-------------------|--------------------------|----------------------|
| 100 kHz        | 1                     | 30               | 1                      | 30                | Y                        | Y                    |
| 160 kHz        | 1                     | 30               | 1.7321                 | 0                 | N                        | N                    |

Table 3. Related parameters in the simulation process

| Data type                        | Value                      |
|----------------------------------|----------------------------|
| frequency of received signal     | 2 MHz                      |
| amplitude of received signal     | 1 V, 0.5 V                 |
| phase of received signal         | 30°, 45°                   |
| frequency of undersampling       | 128 kHz                    |
| frequency after superheterodyne  | 6 kHz                      |

From figure 5, the undersampling technology can effectively recover the amplitude and phase information of the received signals under noise conditions. The feasibility of applying the undersampling technology in actual environment is verified, thus providing a reliable basis for applying the undersampling technology to instruments. Table 4 describes the attenuation, phase shift and corresponding recovery degree obtained by the undersampling technology in detail.

The received signals are also processed through the superheterodyne technology. In the superheterodyne process, the high-frequency signals are reduced to 6 kHz. We set $F_r = 1.994$ MHz, the processing results obtained through the traditional technology are shown in figure 6.

It can be seen from the comparison between the result of figures 5 and 6 that both the undersampling and superheterodyne processing results are similar, indicating that the undersampling technology can be as effective as or even better than the traditional superheterodyne processing. To better analyse and compare the effects of the two methods, the relevant data applied in the simulation process are shown in table 4.

Compared the data in table 4, the attenuation can be recovered to 5.9965 dB and the phase shift can be recovered to $15.0555^\circ$ using undersampling technology. The recovery effect of undersampling technology is similar to that of superheterodyne technology. The simulation shows that the recovery accuracy of undersampling technology reaches that of the traditional method.

In summary, the simulation verifies that equation (17) is correct. The effect of undersampling processing is similar to that of the traditional processing, meeting the accuracy requirements. It provides a solid theoretical basis for the application of the undersampling technology, and ensures that more stable measuring data is obtained. In the simulation process, the same signal-to-noise ratio data is used for both processing methods. In the process of practical application, the simple circuit under the undersampling processing method can provide cleaner test data. The next section provides the comparison of actual application effects of the undersampling and superheterodyne technologies through the measured data.

4. Measured data process

To demonstrate some advantages of the undersampling technology such as simple design circuit, the electromagnetic propagation LWD tool, which was developed by the Institute of Geology and Geophysics, Chinese Academy of Sciences is applied to write different data processing programs into the instrument firmware for testing. The whole test is conducted in a stable environment. The results stability of the undersampling technology and traditional measurement method are compared and analysed. The test process was conducted in the suburbs, where the electromagnetic interference is relatively small.

The test instrument is suspended in the air 10 m away from the ground, ensuring that there is no ferromagnetic substance within 6 m from around it. The test diagram is shown in figure 7, where the instrument is suspended to the air for measurement for approximately 7 min.

Long source distance signals of 2 MHz frequency are used for analysis and comparison, and approximately 100 sets of test data are adopted for testing. The original measured data are shown in the figure 8.

The data are processed into attenuation and phase shift data, which are corrected by the suspending correction (Yang et al. 2012). The standard deviation of the test data is then calculated to reflect the stability of the data.

We can see from figures 9 and 10 that the data processing based on the undersampling technology is more stable than that based on the traditional technology. Comparing the data in table 5, the standard deviation of attenuation and phase shift obtained through the undersampling technology are of order of magnitude $10^{-3}$, while the standard deviation of the processing result obtained through traditional processing method are of an order of magnitude $10^{-2}$.

Owing to the simple circuit structure of the undersampling technology, where the circuit noise interference is reduced, in a stable operating environment the instrument maintained at all times improved measurement accuracy of the instrument.
Figure 5. (a) Rx1 received signal; (b) Rx1 undersampled signal; (c) Rx1 processing results; (d) Rx2 received signal; (e) Rx2 undersampled signal and (f) Rx2 processing results.
Table 4. Comparison of different sampling frequency processing result.

|                     | Initial amplitude (V) | Initial phase (°) | Recovery amplitude (V) | Recovery phase (°) |
|---------------------|-----------------------|-------------------|------------------------|-------------------|
| undersampling       |                       |                   |                        |                   |
| Rx1                 | 1                     | 30                | 0.9967                 | 30.0066           |
| Rx2                 | 0.5                   | 45                | 0.4997                 | 45.0621           |
| ATT or PS           | 6.0206 dB             | 15                | 5.9965 dB              | 15.0555           |
| superheterodyne     |                       |                   |                        |                   |
| Rx1                 | 1                     | 30                | 1.0001                 | 30.0641           |
| Rx2                 | 0.5                   | 45                | 0.5002                 | 44.9958           |
| ATT or PS           | 6.0206 dB             | 15                | 6.0179 dB              | 14.9317           |

Figure 6. (a) Rx1 received signal; (b) Rx1 post-lower-frequency signal; (c) Rx1 processing results; (d) Rx2 received signal; (e) Rx2 post-lower-frequency signal and (f) Rx2 processing results.

The design of the undersampling circuit is relatively simple and because of this it can effectively avoid the interference of circuit noise. We can apply this to processing data, which are measured by electromagnetic propagation LWD. We also can apply this technology to other fields according to the actual situation.

5. Conclusion

This study proposes a data acquisition and processing method based on undersampling technology. This method can reduce the measurement error of data, enhancing stability of data and providing support for accurate formation.
evaluation and reservoir identification. Simple processing flow can reduce the difficulty of firmware writing, which will facilitate instrument research and development.

In this paper, we postulate the undersampling frequency satisfy $F_0/F_s = k/2 (k > 1, k \in \mathbb{Z})$ according to equation derivation. To simplify the firmware writing and reduce the difficulty in subsequent calculations, the signal correction coefficients are developed from analysing the physical meaning of the undersampling technology. The correction coefficients ensure the received signals at different frequencies are undersampled at the same frequency and the recovered signals are same frequency.

In this study, the selection range of the undersampling frequency is verified corrected by simulation. The undersampling technology is proven as feasible according to the comparison of the processing effect of the undersampling with conventional technologies. To better reflect the circuit advantages of the undersampling technology, different data processing methods are used to process the field data. We see

**Figure 7.** Diagram of instrument test environment.

**Figure 8.** (a) The measured attenuation data and (b) phase shift data.

**Figure 9.** Comparison of attenuation stability from undersampling and superheterodyne processing.

**Figure 10.** Comparison of phase shift from undersampling and superheterodyne processing.
from the comparison of the field data processing results that the undersampling technology exhibits a small standard deviation and is more stable. In conclusion, the electromagnetic propagation LWD tool can obtain more stable processing results while using undersampling technology, which makes measurement results more reliable and accurate. The undersampling technology is more suitable for instrument application.

Acknowledgements

This work was supported by the Strategic Priority Research Program of the Chinese Academy of Sciences (grant nos. XDA14020101, XDA14020401); The CAS-CNPC Strategic Cooperation Project (grant no. 2015B-4016); The Youth Innovation Promotion Association of the Chinese Academy of Sciences (grant no. 2019YFA0708301).

Conflict of interest statement. None declared.

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