Channel Sounding in White Space Spectrum
Customer Application Note

Products:

| R&S®SMBV100A

| R&S®FSL

The aim of channel sounding is to characterize a radio channel by decomposing the radio propagation path into its individual multipath components. This application note provided by Neul describes how channel sounding can be performed using the R&S®SMBV100A Vector Signal Generator and the R&S®FSL Spectrum Analyzer. To illustrate the procedure reference is made to 8 MHz wide white space channels in the UK.

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Application Note

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1 Introduction

The aim of channel sounding is to characterize a radio channel by decomposing the radio propagation path into its individual multipath components. This information is essential for developing robust modulation schemes to pass data over the channel.

This application note provided by Neul (www.Neul.com) describes how channel sounding can be performed using the R&S®SMBV100A Vector Signal Generator and the R&S®FSL Spectrum Analyzer. To illustrate the procedure reference is made to 8 MHz wide white space channels in the UK. By simply scaling the R&S®SMBV100A ARB clock frequency and the R&S®FSL sample rate, the procedure can readily be adapted to other channel bandwidths.

In the following, the R&S®SMBV100A Vector Signal Generator will be referred to as SMBV. The R&S®FSL Spectrum Analyzer will be referred to as FSL.

For the measurements described in this application note, the **SMBV** needs to be equipped with one of the two frequency options R&S®SMBV-B103 (9 kHz to 3.2 GHz) or R&S®SMBV-B106 (9 kHz to 6 GHz), and with one of the following baseband generator options:

- R&S®SMBV-B10 with digital modulation (realtime) and ARB (32 Msample), 120 MHz RF bandwidth
- R&S®SMBV-B50 with ARB (32 Msample), 120 MHz RF bandwidth
- R&S®SMBV-B51 with ARB (32 Msample), 60 MHz RF bandwidth

Optionally, the ARB memory of all three baseband generators can be extended to 256 Msample with R&S®SMBV-B55. Use of the hard disk R&S®SMBV-B92 is required with R&S®SMBV-B55 and with R&S®SMBV-B10, but recommended for other configurations as well.

The **FSL** is available with frequency ranges of either 9 kHz to 3 GHz, 6 GHz or 18 GHz. The FSL used in this application note is furthermore equipped with options R&S®FSL-B22 (RF Preamplifier), R&S®FSL-B30 (DC Power Supply, 12 V to 28 V), and R&S®FSL-B31 (NiMH Battery Pack).

See chapter 7 for details on ordering information.

Stationarity

Real channels are not stationary but vary with time. This may be due to movement of the transmitter and/or receiver or it may be caused by movement of the objects responsible for the multipath components. The analysis presented here concentrates on stationary propagation paths but could readily be extended to slow time variations.

White Space

The retirement of analogue TV transmitters and their replacement with DVB-T transmitters has led to the freeing up of previously occupied spectrum. It is likely that some of this spectrum will become available for new communication systems on a worldwide basis. This spectrum is referred to as white space.
In the UK the white space spectrum is divided into 8 MHz channels, as illustrated in Figure 1. It is indicated as “retained / interleaved spectrum” in Figure 1, as this spectrum can be shared between white space devices and digital terrestrial television.

In order to design communications systems to exploit this spectrum, knowledge of the characteristics of the radio propagation paths is required. This application note describes how channel sounding of these 8 MHz channels has been performed using the FSL and SMBV. Please check with your local regulatory authority for license requirements before performing the tests.

Figure 1: White space spectrum in UK (PMSE = Programme-Making and Special Events)

2 Principles of Channel Sounding

Single impulse method

The simplest method of channel sounding would be to transmit a band limited impulse and record the signal which arrives at the receiving location, see Figure 2.

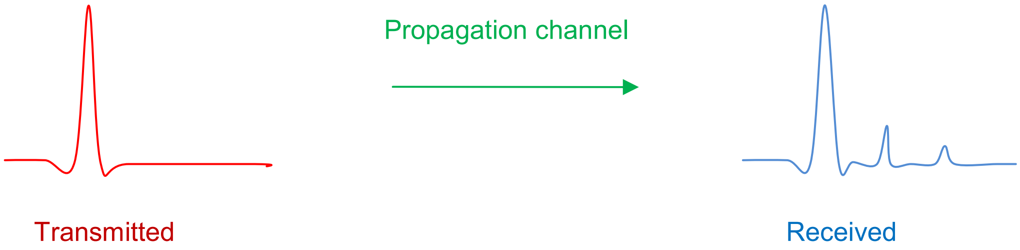

Figure 2: Single impulse method

In practice this method is not viable. The magnitude of the impulse which can be transmitted will typically be limited due to the license requirements for performing such tests (please check with your local regulatory authority). This is unlikely to permit more power than that required to establish a communications link between transmit and receive locations, which may be designed to work at a negative signal-to-noise at the largest ranges. Hence the received impulse would probably be buried in noise.

Sequence of impulses method

The situation can be improved by transmitting a sequence of pulses, see Figure 3. The individual pulses can then be averaged at the receiver to improve the signal-to-noise ratio. This method relies on:

– The separation of the impulses in time being greater than the longest multipath delay.
– The channel being stationary over the averaging period.

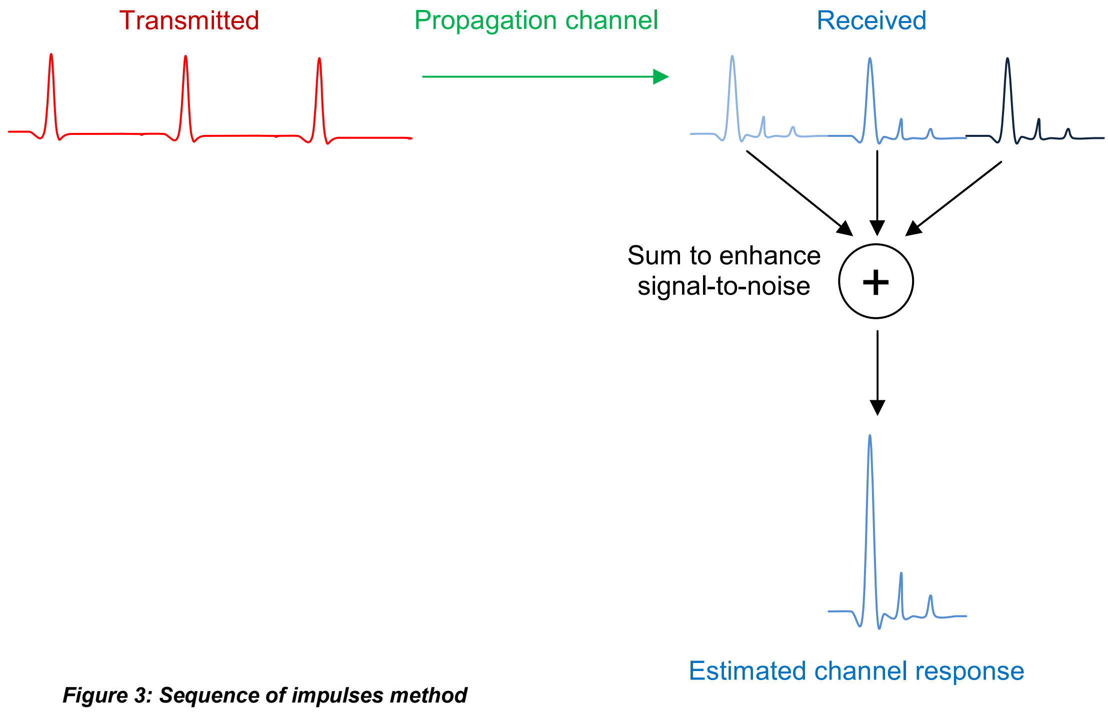

Figure 3: Sequence of impulses method

Although an improvement over the single impulse method, the sequence of impulses is still not viable. The remaining problem is that the power is concentrated in time; the peak-to-average power ratio for the transmitter is excessive.

Fourier decomposition method

Some means of spreading the transmitted energy in time must be found. There are numerous methods for doing this, for example chirps are commonly used. The method described here is believed to have a small novel component and is simple to implement.
The sequence of impulses can be Fourier decomposed into a set of discrete tones. The phases of the tones are then adjusted to spread the energy in time before being recombined to generate a repetitive transmit waveform. At the receiving end the waveform is decomposed back into the individual tones, the inverse phase rotations are applied and the tones recombined to yield the channel impulse response function.

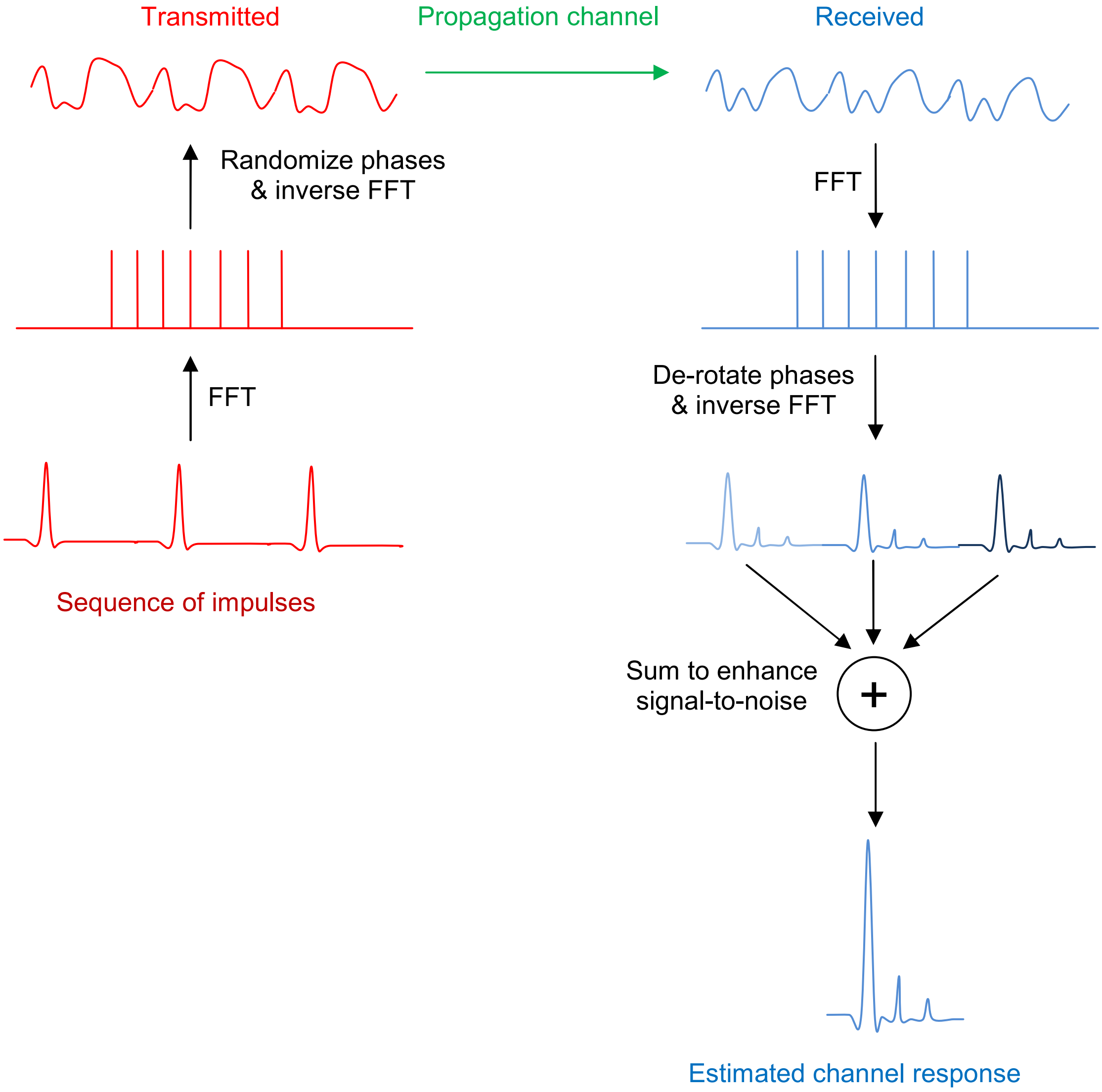

Figure 4: Fourier decomposition method

3 Implementation of Channel Sounding

Selection of waveform repeat period

The waveform used is composed of a number of tones uniformly separated in frequency. The tone spacing determines the repeat period of the waveform. The repeat period of the waveform is an important parameter:

– The longest multipath delay which can be resolved is equal to the waveform repeat period. If the repeat period is too short then the estimated channel impulse response will be confused.

– The channel will only remain stationary over some finite time interval. The number of waveform repeats which fit in this interval determines the maximum averaging and hence the sensitivity of the channel sounding. Therefore smaller waveform repeat intervals lead to greater sensitivity.

A typical white space channel does not contain significant multipath components at delays longer than 30 µs. For stationary transmit and receive locations the channel is likely to remain constant for tens of milliseconds. Neul typically use a 64 µs waveform repeat which in an 8 MHz channel contains 257 tones. At a sample rate of 16 MSps the FSL can record 32 ms of data which is approximately equal to the length of time over which the channel is stationary.

Selection of tone phase rotations

The novel part of the channel sounding method described in this application note is the selection of the tone phase rotations. This is not dissimilar from techniques used for peak-to-average power reduction in OFDM systems. The scheme iterates between the time domain and Fourier domain to gradually reduce the peaks in the time domain whilst still maintain the tones at constant amplitude in the Fourier domain. The method is illustrated by the block diagram in Figure 5. Example Matlab code can be found in the Annex of this document.

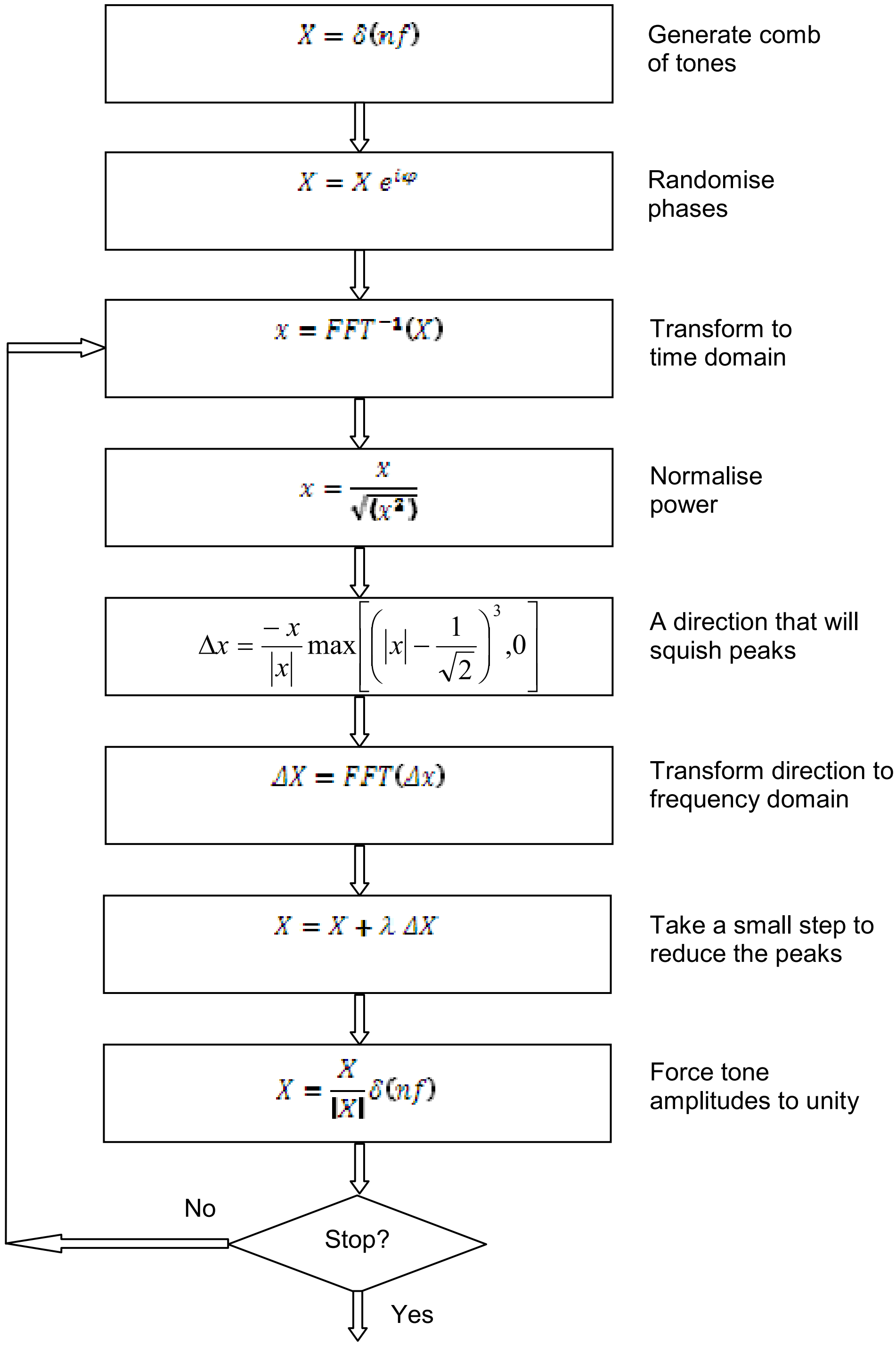

Figure 5: Block diagram of used channel sounding method

This method will typically yield a waveform with a peak-to-average power ratio around 2 dB.

Setting up the SMBV

The resulting waveform can be loaded into the ARB of an SMBV vector signal generator. The ARB clock rate should be set to twice the channel bandwidth since the waveform is oversampled to ensure that the peaks are correctly suppressed.
The following sequence of commands will load the waveform into the ARB, set the frequency to channel 37 (602 MHz), and generate an average output power of around 20 dBm.

```
*RST
BB:ARB:WAV:DATA '/var/user/tones_256.wv' {TYPE: SMU-WV, 0}{DATE:
2011-06-05;07:44:05}{LEVEL OFFS: 0.000000,0.000000}{SAMPLES:
512}{CLOCK: 16000000}{WAVEFORM-2049: #...
BB:ARB:WAV:SEL '/var/user/tones_256.wv'
FREQ:FIX 602000000 Hz
POW:AMPL 22.5 dBm
OUTP:STAT ON
```

With the constrained peak-to-average power ratio substantial output energy can be obtained directly from the SMBV eliminating the need for an additional power amplifier. Figure 6 illustrates a 256 tone, 8 MHz wide waveform used for white space channel sounding. This was recorded from an SMBV outputting 20 dBm. Even at these high output powers the spurious emissions in the adjacent channels are well suppressed.

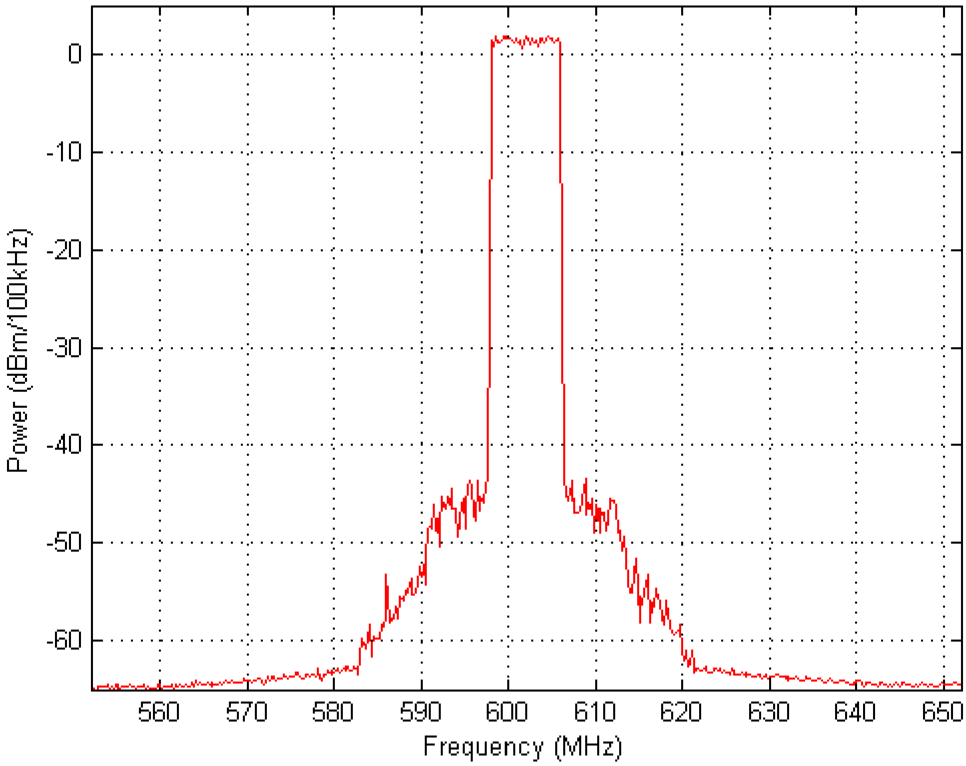

Figure 6: Spectrum of waveform used for white space channel sounding

Reception of waveform

The FSL is an ideal instrument for the reception of the channel sounding waveform. For white space channels it has sufficient memory to hold in excess of 30 ms of data. This is sufficiently long that time variations in the channel start to become apparent. In addition, it can be powered from an internal battery allowing it to be used in the field.
To simplify the processing of the received waveform, it is recorded at twice the channel bandwidth, i.e. at the same sampling rate as the waveform loaded into the SMBV ARB.
The following sequence of commands will set the FSL to channel 37 (602 MHz), adjust the span to encompass the wanted signal and turn the preamplifier (option FSL-B22) on:

```
*RST
SENS:FREQ:CENT 602000000 Hz
SENS:FREQ:SPAN 16000000 Hz
SENS:BAND:RES 100000 fHZ
INP:GAIN:STAT 1
```

At this stage it is best to adjust the reference level on the analyzer manually. Keep an eye on the overload indicator whilst doing this. The remaining commands to retrieve 32 ms of data are:

```
TRAC:IQ:STAT ON
TRAC:IQ:SET NORM,16000000HZ,16000000HZ,IMM,POS,0,512000
FORMAT REAL,32
TRAC:IQ:DATA?
```

Determination of frequency error

The first stage in processing the received waveform is to correct for the frequency error between the SMBV and FSL. This could be done using one of the numerous frequency error estimation schemes which are found in communication systems. However, since the frequency error is small and processing time is unlikely to be critical, a brute force search has been adopted.
Figure 7 shows a block diagram for frequency error correction. Example Matlab code can be found in the Annex. It corrects the received data by a postulated frequency error. The frequency corrected data is then correlated with the transmitted waveform and the peak of the correlation found. The postulated frequency error which yields the largest correlation peak is used for subsequent analysis.

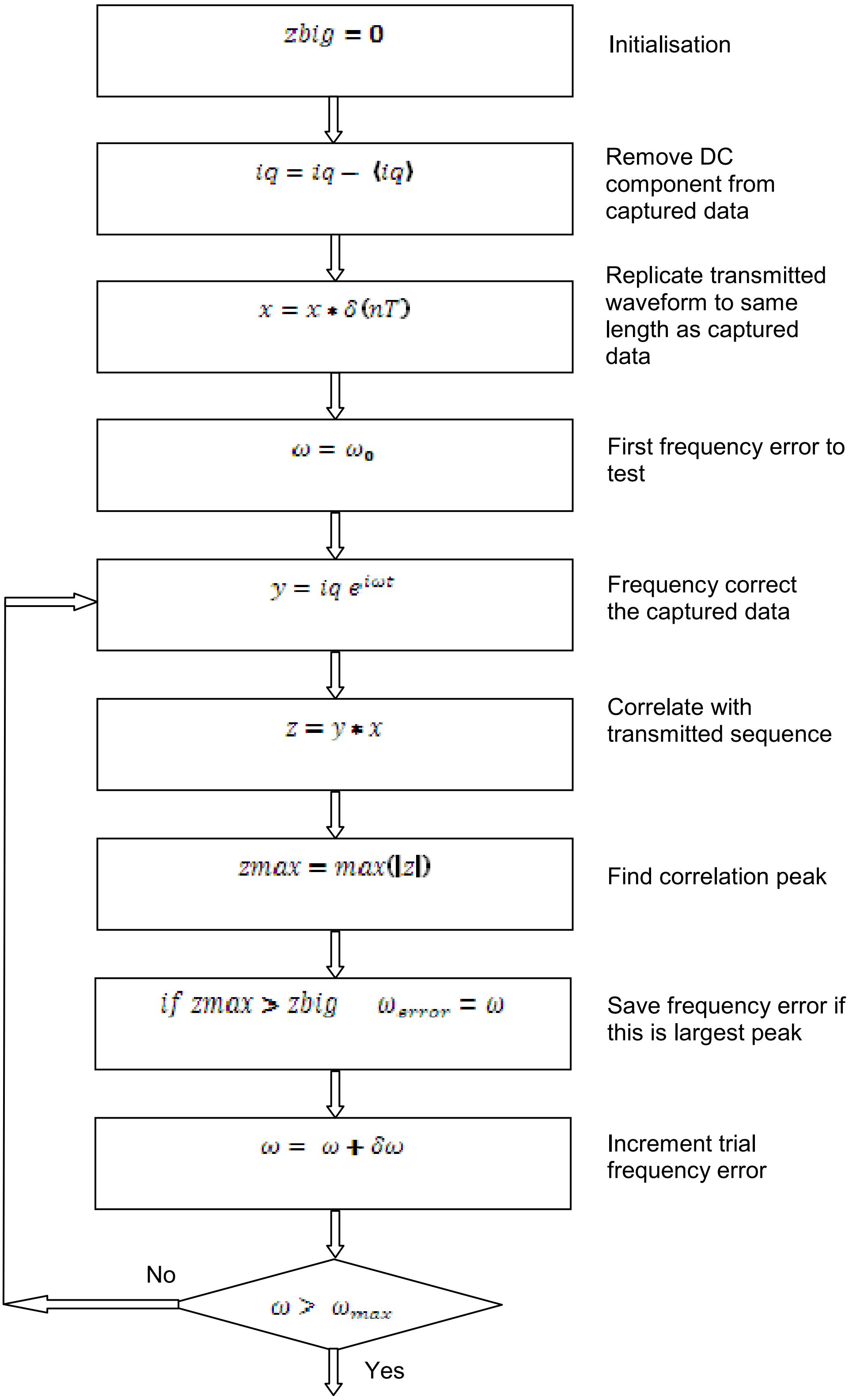

Figure 7: Block diagram for frequency error correction

Running the example Matlab code on captured IQ data will generate a graph showing the estimated frequency error, see Figure 8. The red points correspond to the first coarse search and the blue points to the second refined search. This scheme is simple, reliable and will work with negative signal-to-noise ratios.

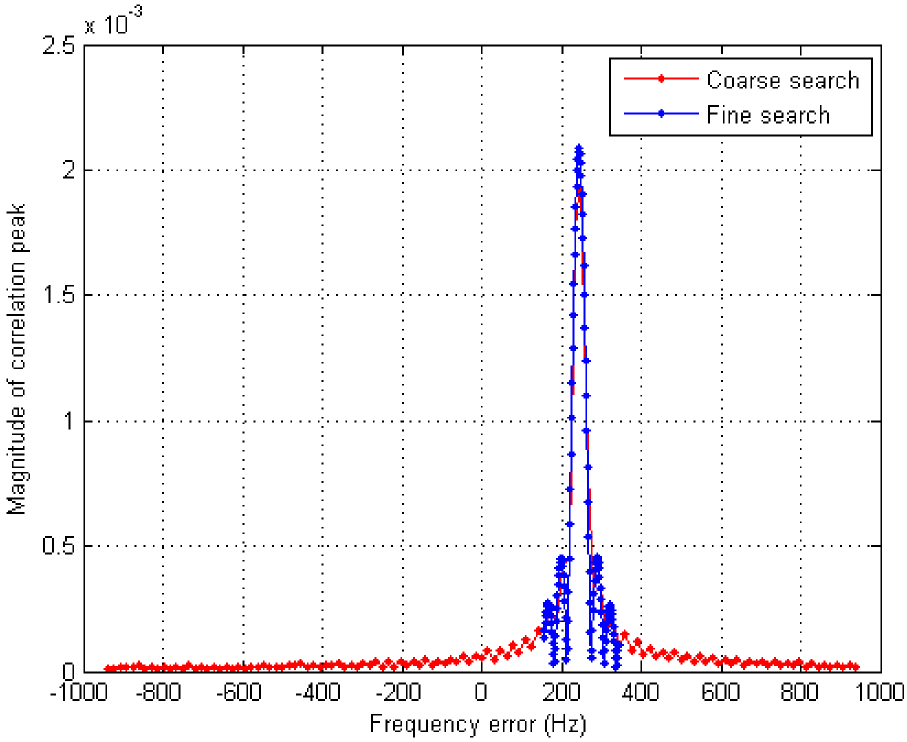

Figure 8: Frequency error correction

Determination of channel impulse response

Having removed the frequency error between the SMBV and FSL, the channel impulse response could be estimated by Fourier decomposing the received waveform, de-rotating the transmitted tones and then transforming back to the time domain. Figure 9 shows the principle in a block diagram. The Matlab example code (see Annex) actually uses a Wiener filter to estimate the channel impulse response which may give more reliable results in low signal-to-noise ratio scenarios.

$$XX = FFT(xx)$$ $$IQ = FFT(iq)$$	Transform to the frequency domain
$$E_{signal} = \langle |IQ|^2 \delta(nf) \rangle$$	Calculate mean energy in received tones
$$E_{noise} = \langle |IQ|^2 (1 - \delta(nf)) \rangle$$	Calculate mean energy between received tones
$$SNR = \frac{E_{signal}}{E_{noise}}$$	Estimate signal-to-noise ratio of tones
$$\gamma = \frac{\langle |XX|^2 \rangle}{SNR}$$	Calculate γ factor
$$H = \frac{IQ\ XX^*}{|XX|^2 + \gamma}$$	Apply Wiener filter
$$\boldsymbol{h} = FFT^{-1}(H)$$	Transform back to time domain

Figure 9: Block diagram for determination of channel impulse response

Sensitivity

As with any spectrum analyzer, the noise floor of the FSL is not the same as can be achieved with a dedicated LNA. Hence there may be some occasions in which an external LNA is required. However, the sensitivity of the channel sounding system using only the internal FSL preamplifier (option FSL-B22) is quite astounding.
To test the sensitivity a simple multipath model was constructed, see Figure 10. It consisted of a small pre-cursor signal followed by low amplitude long delay multipath. This is illustrated below. The pre-cursor gives rise to a long periodic wobble in the frequency domain whilst the post-cursor imposes a short period ripple.

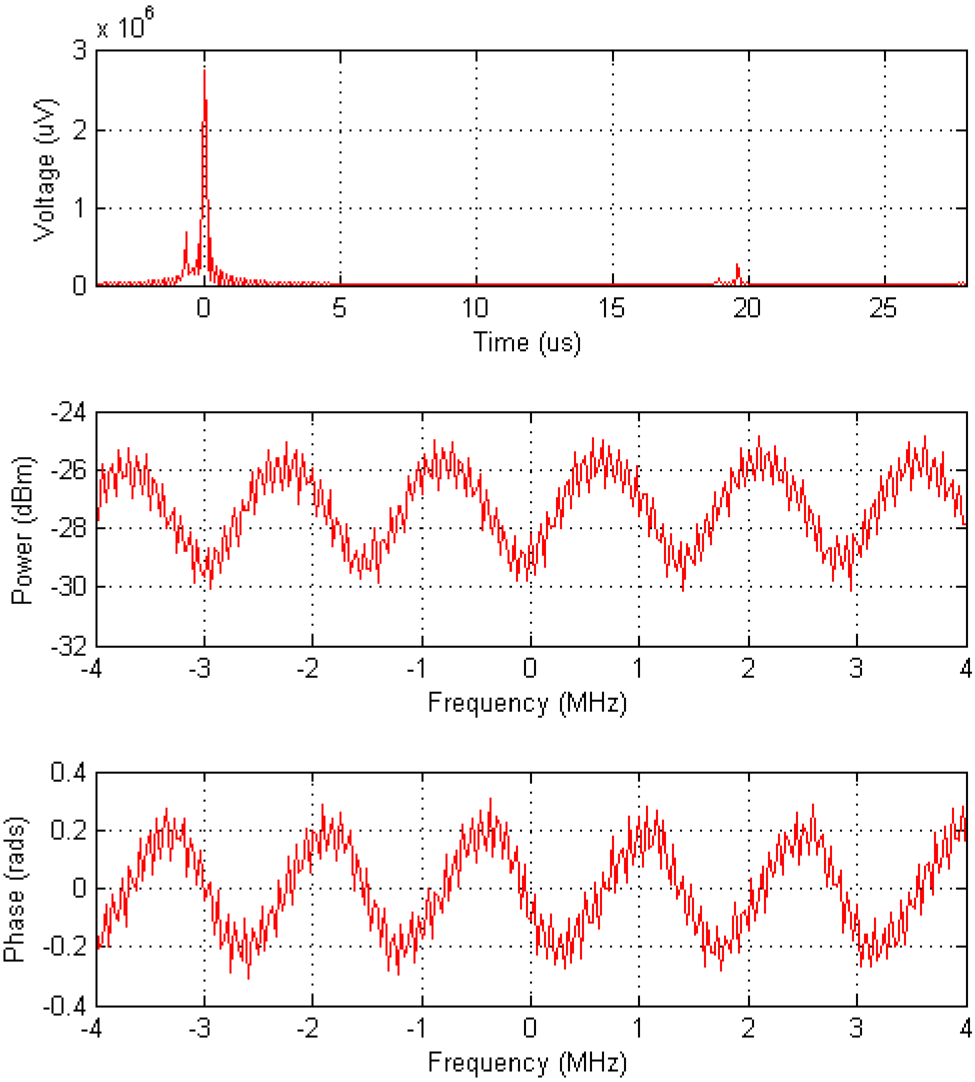

Figure10: Multipath model

The signal generator level was reduced until the signal entering the FSL was at -110 dBm in an 8 MHz white space channel. Even at this level the structure of the simulated channel response is clearly visible, see Figure 11.

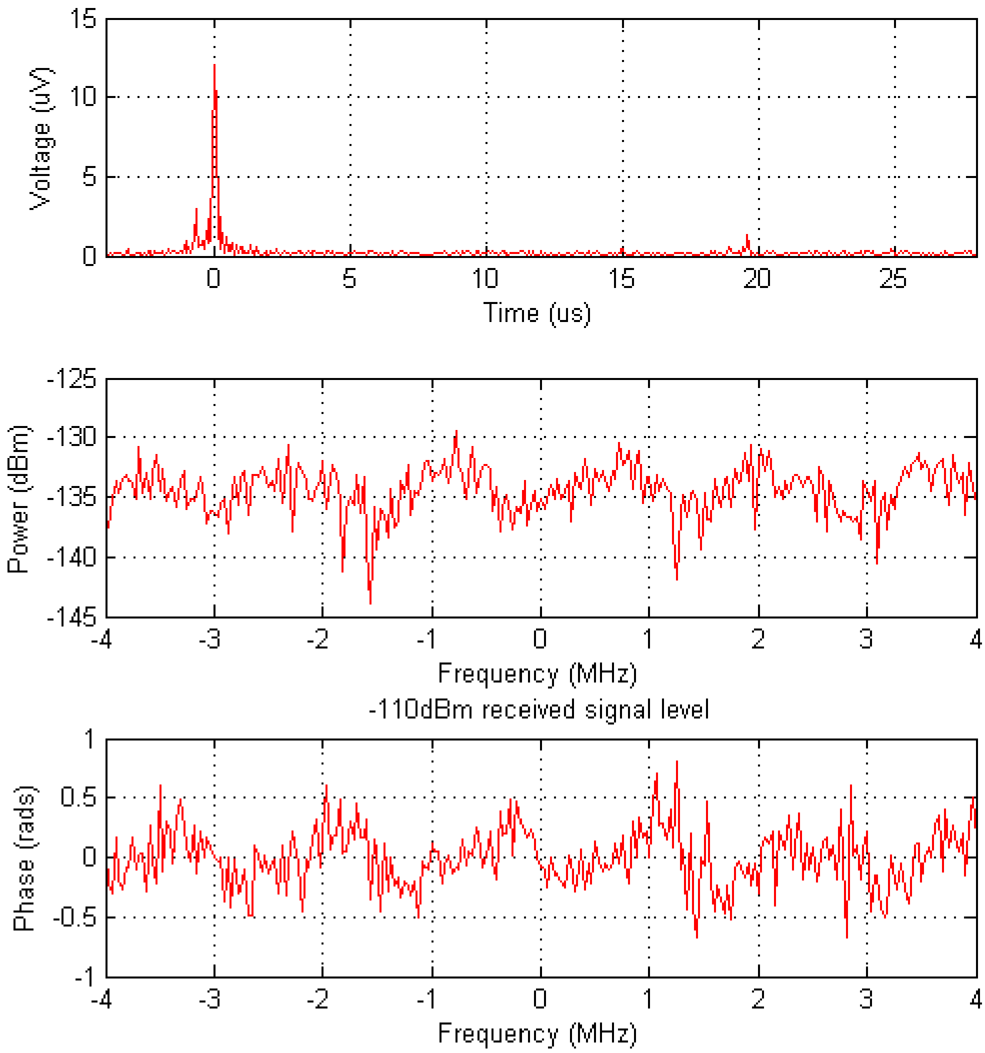

Figure 11: Impact of reduced receive signal level

Since the SMBV can readily achieve a transmit level of +20 dBm, this measured sensitivity of the FSL shows that channel sounding with propagation losses of order 130 dB is entirely feasible without an external LNA. In practice both transmit and receive antennas are likely to have some gain, this is typically 4 dBi for Neul antennas. Use of these antennas extends the propagation loss that can be tolerated to 138 dB. Use of an external LNA permits signals as low as -125 dBm to be resolved. In combination with the SMBV's +20 dBm output power and 4 dBi gain antennas, this permits a path with propagation loss in excess of 150 dB to be sounded.
Although the FSL is unable to resolve detail in the channel response much below -110 dBm, it can still reliably find and measure the main path. The plots in Figure 12 shows the system still functioning at -133 dBm without an external preamplifier. With 20 dBm SMBV output power and 4 dBi antennas, this would correspond to a total path loss in excess of 160 dB.

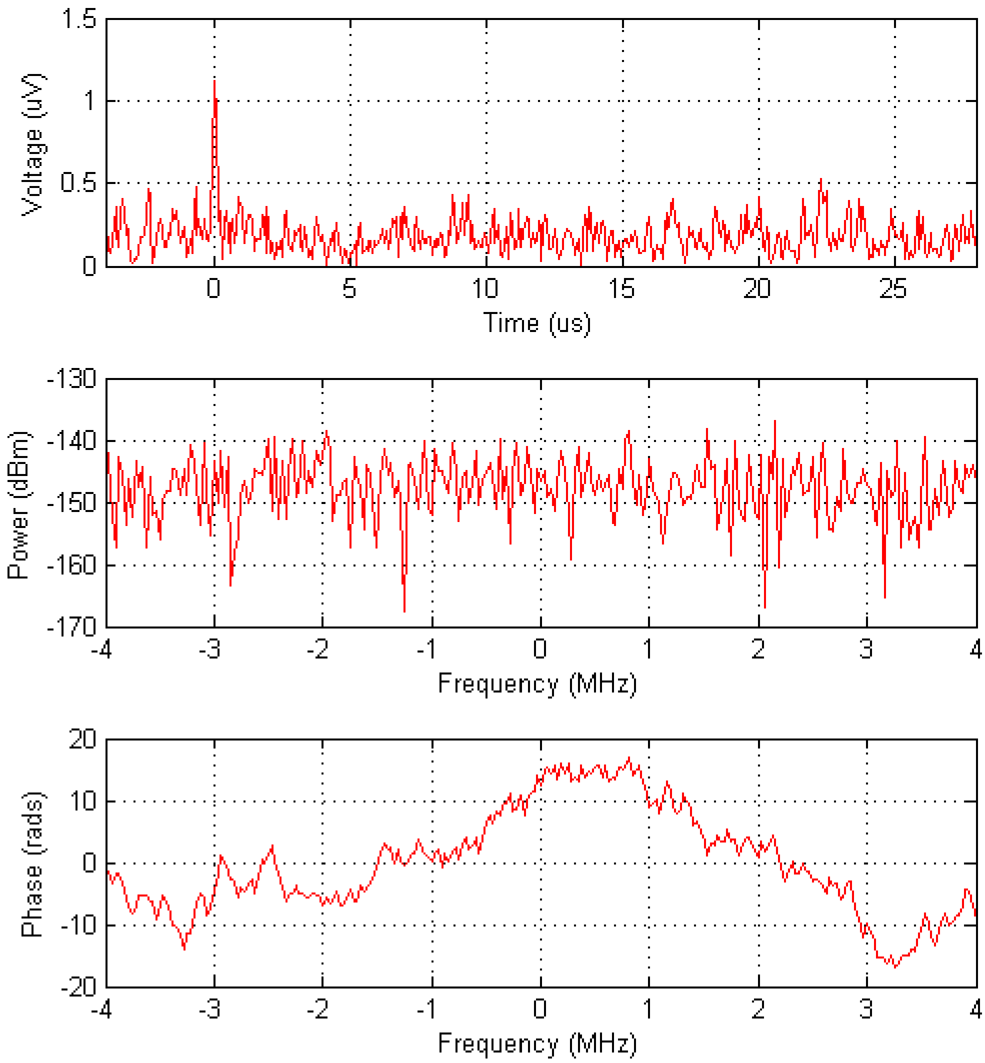

Figure 12: Impact of further reduced receive signal level

4 Examples of Real White Space Channels

This section presents examples of two white space channels which have been sounded by Neul using the SMBV and FSL. The first is an interesting sub-urban path whilst the second is unique to the architecture of Cambridge.

Riverside to Industrial Estate

In this experiment the transmit antenna was located about 6 m above ground level on the banks of the river Cam. Being in a river valley the transmit location was far from ideal. The receive location was 1.5 km away in the middle of an industrial estate. The channel impulse response in Figure 13 shows a number of similar amplitude paths extending over 2 µs. These multipaths are thought to originate from the metal cladding of the large warehouses within the industrial estate.

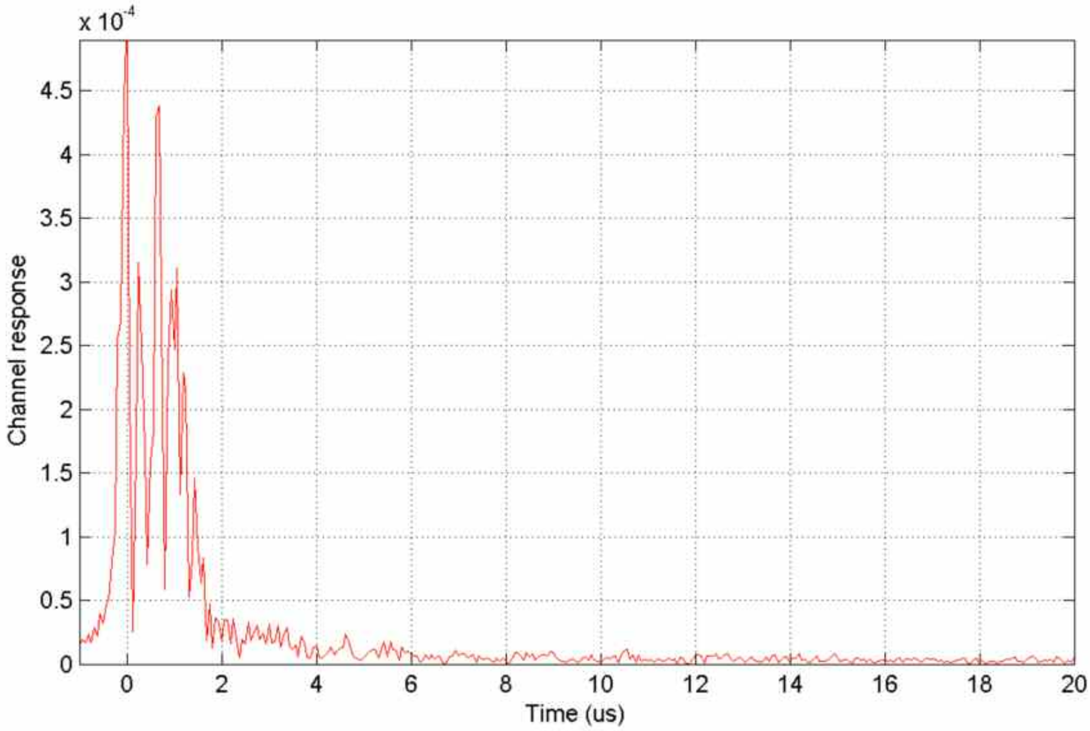

Figure 13: Channel sounding result in sub-urban environment

Gonville and Caius College

The propagation channels between various rooms within Gonville and Caius College were measured using the SMBV and FSL over distances less than 100 m. None of these paths were line of sight. In fact, numerous stone walls lay between transmit and receive locations. These walls are substantial, some of them being in existence since 1353. Penetration of the walls by white space radio is thought to be unlikely (students frequently complain about the lack of mobile phone coverage within the college) and it is suspected that many of the paths observed were due to multiple diffractions. A typical channel is shown below. There are a number of closely spaced paths which have a delay spread of order 400 ns. This is surprisingly large since the travel time for a direct path (if one were to exist) would only be 230 ns. This suggests that some of these paths have arrived by quite exotic routes.

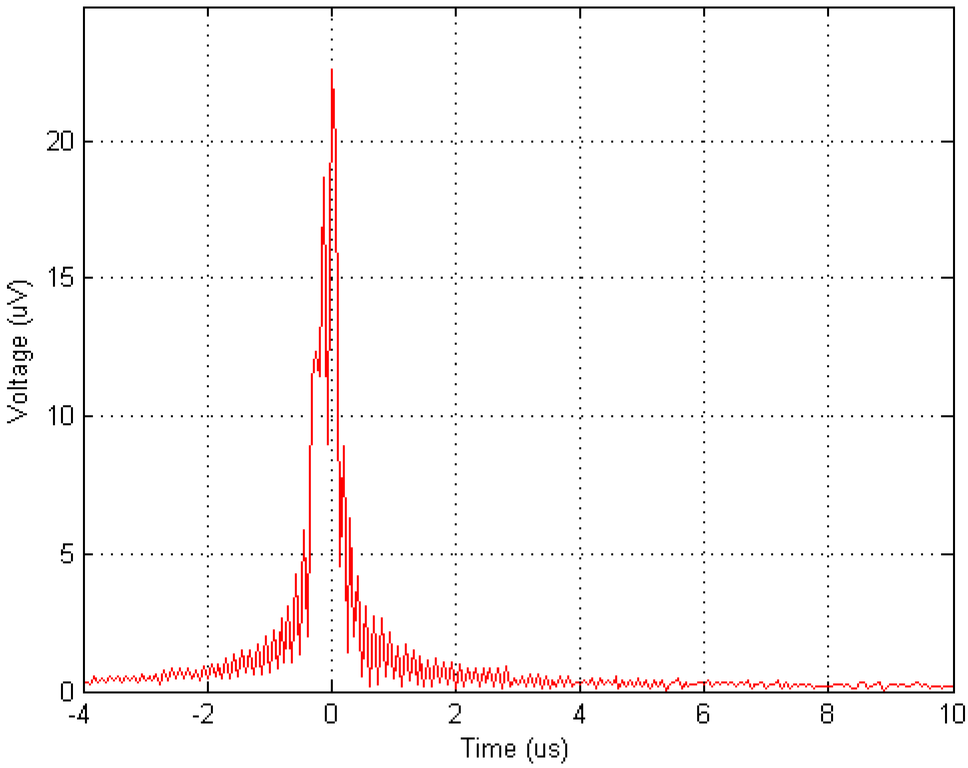

Figure 14: Channel sounding result in historic urban architecture

5 Annex: Example Matlab Code

5.1 Example Matlab code illustrating the used channel sounding method

```
% N is the number of tones to be transmitted which must be odd
% Suggested values for N are 2^n + 1
N = 257;
M = (N+1)/2;

% Start by assigning random phases to the tones
Phi = 2*pi*rand(1,N);
% Generate X which is the Fourier transform of the waveform
X = exp( 1i * Phi );
% Pad with zeros so that the time domain waveform will be 2x
% oversampled
X = [ X(1:M) zeros(1,N-2) X(end-M+2:end) ];

% Lambda is the step size per iteration
Lambda = 0.001;

% An iterative loop during which the phases converge to the
% wanted values
for Iter = 1 : 20000

    % Transform the waveform back to the time domain
    x = ifft( X );

    % Normalise the power in the waveform
    x = x / sqrt( mean( abs(x).^2 ) );

    % Calculate a vector which will push the peaks of the
    % waveform towards the mean
    gradx = - ( x ./ abs(x) ) .* max( (abs(x) - 1/sqrt(2)).^3 ,
0 );

    % Transform the vector into the Fourier domain
    GradX = fft( gradx );

    % Update the Fourier transform of the waveform to reduce the
    % peak to average power ratio
    X = X + Lambda .* GradX;

    % Force the Fourier components back to unity amplitude
    X = X ./ abs(X);
    % Zero the Fourier bins which do not contain tones
    X(M+1:end-M+1) = 0;

end
```

5.2 Example Matlab code for frequency error correction

```
% iq holds the data captured from the FSL
M = length(iq);
% Ntx is the number of samples in the transmitted waveform
% The transmitted waveform is oversampled by a factor of 2
Ntx = 2 * 256;

% Work out how many complete repeats of the transmitted waveform
have been
% captured
% nbasic is the length of the transmitted waveform
Nrep = floor( M / Ntx );
% Truncate the captured iq data to be a complete number of
% waveform repeats
M = Ntx * Nrep;
iq = iq(1:M);

% Remove the DC component since this will be in error at
% negative signal-to-noise ratios
iq = iq - mean(iq);

% Replicate the waveform held in the SMBV ARB so that it is the
% same length as the data captured from the FSL
% x is the waveform which was loaded into the SMBV ARB
xx = repmat( x , 1 , Nrep );
% Normalise the power in the replicated waveform
xx = xx * / sqrt( sum( abs(xx).^2 ) );
xx = xx(:).';
% Convert replicated waveform to Fourier domain and take its
% conjugate
XX = conj( fft(xx) );

% Frequency search over +/- 1kHz
% For 32ms of captured data 1kHz is roughly 30 cycles
Cycles = -30:0.5:30;
Foffset = 0;

figure(1);
clf;

% The 8MHz channel is 2x oversampled by the FSL
% dt is the time per sample
dt = 1 / 16e6;
% Ttotal is the duration of the captured data
Ttotal = M * dt;
% Precalculate the time vector
t = 1i * 2 * pi * ( 0 : M-1 ) / M;

% The estimation of the frequency offset is done in two stages
% The first stage scans from -1kHz to +1kHz in increments of
% half a cycle over 32ms
% The second stage does a refined search around the peak found
% in the first stage
```

```
for Iter = 0 : 1

    Peaks = [];

    % Loop over possible frequency errors
    for Ncycle = Cycles

        % Correct the captured data by the postulated frequency
        % error
        y = iq .* exp( Ncycle * t );
        % Correlate with the SMBV ARB waveform
        z = ifft( fft(y) .* XX ) )
        % Save the correlation peak value
        Peaks(end+1) = max( abs( z ) );

    end

    % Plot the correlation peaks
    if( Iter )
        plot( Foffset + Cycles/Ttotal , Peaks , '.-b' );
    else
        plot( Foffset + Cycles/Ttotal , Peaks , '.-r' );
    end
    hold on
    grid on
    drawnow;

    % Find the maximum correlation peak
    [~,k] = max(Peaks);
    % How many cycles did this correspond to?
    Ncycle = Cycles(k);
    % Correct the captured data by the estimated frequency
    % offset
    iq = iq .* exp( Ncycle * t );

    % Update the estimate of the frequency offset
    Foffset = Foffset + Ncycle/Ttotal;

    % Reduce the search range for the second stage
    Cycles = Cycles / 10;

end

xlabel( 'Frequency offset (Hz)' );
ylabel( 'Correlation' );
```

5.3 Example Matlab code for determination of the channel impulse response

```
% Transform the frequency corrected data to the Fourier domain
IQ = fftshift( fft(iq) );
% Transform the replicated transmit waveform to the Fourier
% domain
XX = fftshift( fft(xx) );

% Find the indices in the FFT of the tones - crude but simple
% Find the magnitude of the tones
Xmax = max( abs(XX) );
% Locate those Fourier bins which do not contain tones
XX_no  = find( abs(XX) <= 0.1 * Xmax );        % bins not
containing tones
% Locate those Fourier bins which do contain tones
XX_yes = find( abs(XX) >  0.1 * Xmax );        % bins containing
tones
% Find range of Fourier bins which contain tones
Xlo = min(XX_yes);
Xhi = max(XX_yes);
% Find bins which lie between the tones
k = find( (XX_no > Xlo) & (XX_no < Xhi) );
XX_between = XX_no(k);
% bins between tones

% Calculate mean energy in the bins containing tones
% This is signal energy
Esignal = mean( abs(IQ(XX_yes)).^2 );
% Calculate mean energy in the bins lying between the tones
% This is noise energy
Enoise = mean( abs(IQ(XX_between)).^2 );
% Calculate the signal-to-noise ratio of the tones
SNR = Esignal / Enoise;

% Use a Wiener filter to estimate the impulse response function
Gamma = mean( abs(XX).^2 ) / SNR;
H = IQ .* conj(XX) ./ ( abs(XX).^2 + Gamma );
h = ifft( fftshift(H) );
% Remove all the nasty scaling factors to get back to units of
% Volts
h = h * sqrt(2) * sqrt(Nrep);
```

6 Additional Information

This Application Note has been kindly provided by Neul (www.Neul.com).

This Application Note is updated from time to time. Please visit the website http://www.rohde-schwarz.com/appnote/1MA199 in order to download the latest version.

Please send your comments and suggestions regarding this application note to TM-Applications@rohde-schwarz.com.

7 Ordering Information

Signal generator

R&S®SMBV100A	Vector Signal Generator	1407.6004.02
R&S®SMBV-B103	RF Path 9 kHz to 3.2 GHz	1407.9603.02
R&S®SMBV-B106	RF Path 9 kHz to 6 GHz	1407.9703.02
R&S®SMBV-B10	Baseband Generator, ARB 32 Msample (120 MHz, realtime)	1407.8607.02
R&S®SMBV-B50	Baseband Generator, ARB only, 120 MHz	1407.8907.02
R&S®SMBV-B51	Baseband Generator, ARB only, 60 MHz	1407.9003.02
R&S®SMBV-B55	Memory Extension for ARB, 256 MS	1407.9203.02
R&S®SMBV-B92	Hard Disk	1407.9403.02

Spectrum analyzer

R&S®FSL3	9 kHz to 3 GHz	1300.2502.03
R&S®FSL6	9 kHz to 6 GHz	1300.2502.06
R&S®FSL18	9 kHz to 18 GHz (overrange 20 GHz)	1300.2502.18
R&S®FSL-B22	RF Preamplifier (3 GHz/6 GHz)	1300.5953.02
R&S®FSL-B30	DC Power Supply, 12 V to 28 V	1300.6308.02
R&S®FSL-B31	NiMH Battery Pack	1300.6408.02

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