**Audio-Band Wireless Channel Measurement and Synchronisation for an OFDM Laptop Modem**  
*(University dissertation template – May 17 2025)*

**Abstract**

Accurate characterisation of the acoustic wireless channel between a laptop loud-speaker and the built-in microphone is a prerequisite for reliable file transmission in the IIA GF3 Audio-Modem project. This dissertation surveys, designs and analytically evaluates measurement techniques able to recover the room impulse response (RIR) with sub-sample resolution, discusses their integration into an orthogonal frequency-division multiplexing (OFDM) modem, and addresses the associated time/frequency synchronisation tasks. Detailed implementation procedures, MATLAB/Python style algorithms, and engineering trade-offs are provided, supported by state-of-the-art literature.

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**1 Introduction**

**1.1 Motivation**

The GF3 brief mandates an *offline* audio modem interoperable across student groups and robust to severe multipath echo. A well-resolved channel model h[n]h[n] enables matched-filter reception and frequency-domain equalisation, directly translating into higher data recovery rates.

**1.2 Objectives**

* Compare time- and frequency-domain sounding signals for laptop loud-speaker/microphone links.
* Provide step-by-step implementation recipes that respect project constraints (48 kHz PCM, no external hardware).
* Quantitatively analyse estimator bias/variance and synchronisation sensitivity.

**1.3 Document organisation**

Section 2 introduces channel and OFDM theory; Sec. 3 presents measurement methods; Sec. 4 derives estimators; Sec. 5 details synchronisation; Sec. 6 outlines an experimental set-up; Sec. 7 summarises engineering trade-offs; Sec. 8 concludes.

**2 Physical and Mathematical Background**

**2.1 Acoustic Multipath Channel**

In baseband discrete time the received sample

y[n]=∑l=0L−1h[l] x[n−l]+w[n]y[n]=\sum\_{l=0}^{L-1}h[l]\,x[n-l]+w[n]

models an FIR of length LL plus additive noise w[n]w[n]. For a 5×8 m5\times8\text{ m} room L ⁣> ⁣500L\!>\!500 taps at 48 kHz are typical (reverberation time T60 ⁣≈ ⁣0.4T\_{60}\!\approx\!0.4 s).

**2.2 OFDM Primer**

By stuffing complex QAM symbols XkX\_k into an NN-point IFFT, adding a cyclic prefix (CP) of Ncp≥L−1N\_{\text{cp}}\ge L-1 and exploiting circular convolution, each sub-carrier sees a flat-fading gain CkC\_k after FFT:

Yk=CkXk+Zk.(k=0…N–1)Y\_k=C\_k X\_k + Z\_k. \quad \text{(k=0…N–1)}

([melaudia.net](https://www.melaudia.net/zdoc/comparisonMesure.PDF?utm_source=chatgpt.com))

**3 Channel-Sounding Methodologies**

| **Method** | **Core idea** | **Key equations** | **Pros** | **Cons** |
| --- | --- | --- | --- | --- |
| **Impulse / Dirac** | Play single sample ‘click’ | h[n]≈y[n]h[n]\approx y[n] | Conceptually simple | Requires high SPL, low SNR |
| **Maximum-Length Sequence (MLS)** | Correlate recorded signal with PN-sequence | h=IFFT ⁣{Y[k]X[k]}h=\text{IFFT}\!\bigl\{\frac{Y[k]}{X[k]}\bigr\} | Fast (FFT-based), low crest-factor | Sensitive to loud-speaker non-linearity ([SCIRP](https://www.scirp.org/reference/referencespapers?referenceid=2329634&utm_source=chatgpt.com)) |
| **Exponential / Linear Chirp Sweep** | Deconvolve log-sweep | h[n]=F−1 ⁣{Y/X ⁣∗}h[n]=\mathcal{F}^{-1}\!\{Y/X^{\!\*}\} | High dynamic-range, harmonic rejection ([SCIRP](https://www.scirp.org/reference/referencespapers?referenceid=454885&utm_source=chatgpt.com)) | Longer measurement time |
| **Pilot-OFDM Preamble** | Insert known OFDM block & do LS | C^k=Yk/Xk\hat{C}\_k=Y\_k/X\_k | Naturally embedded in modem | Lower resolution than long sweeps |

**3.1 Implementation Steps (example: MLS)**

1. **Sequence generation**  
   *Choose m=14⇒P=214 ⁣− ⁣1=16383m=14\Rightarrow P=2^{14}\!-\!1=16383*; map {0,1}→{±1}\{0,1\}\to\{\pm1\}.
2. **Oversample** to audio Fs (48 kHz).
3. **Playback & recording** using e.g. *sounddevice* (Python).
4. **Synchronise captures** (see Sec. 5).
5. **Circular deconvolution** via FFT – removes noise-whitened by averaging KK repetitions.
6. **Window** shortest causal part; optionally fit IIR model using Prony-LS.

**3.2 Engineering Metrics**

* **Energy-to-Noise Ratio (ENR)** of sounding signal

ENR=10log⁡10∑∣x[n]∣2N0B\text{ENR}=10\log\_{10}\frac{\sum|x[n]|^{2}}{N\_0B}

* **RMS delay spread**  
  στ ⁣= ⁣∑h[l] (lTs)2∑h[l]−μτ2\sigma\_\tau\!=\!\sqrt{ \tfrac{\sum h[l]\,(lT\_s)^2}{\sum h[l] } - \mu\_\tau^2 }.
* **Coherence bandwidth** Bc≈1/(50στ)B\_c\approx1/(50\sigma\_\tau).

**4 Channel Estimation and Equalisation in OFDM**

**4.1 Least-Squares (LS)**

C^LS=(XHX)−1XHY\hat{\mathbf{C}}\_{\text{LS}}=(\mathbf{X}^\mathrm{H}\mathbf{X})^{-1}\mathbf{X}^\mathrm{H}\mathbf{Y}

With pilot-only diagonal X\mathbf{X}, LS reduces to Yk/XkY\_k/X\_k at pilot tones; interpolate across data tones with piece-wise linear or DFT-based filters.

**4.2 Minimum-MSE (MMSE)**

C^MMSE=RCY (RYY+σw2I)−1Y\hat{\mathbf{C}}\_{\text{MMSE}}=\mathbf{R}\_{CY}\,(\mathbf{R}\_{YY}+ \sigma\_w^2\mathbf{I})^{-1}\mathbf{Y}

Requires a priori delay-spread model; offers 2–3 dB gain over LS for Eb/N0<10E\_b/N\_0<10 dB.

**4.3 Frequency-Domain Equalisation**

The ZF equaliser outputs Xk~=Yk/C^k\tilde{X\_k}=Y\_k/\hat{C}\_k. To mitigate noise enhancement when ∣C^k∣≪1|\hat{C}\_k|\ll1, apply MMSE or water-filling power loading.

**4.4 Matched Filtering**

In time domain: z[n]=∑h∗[L−1−l] y[n−l]z[n]=\sum h^\ast[L-1-l]\,y[n-l]. Providing SNR gain equal to channel taps LL, at the cost of twice the filter length delay.

**5 Synchronisation for Laptop Audio Links**

**5.1 Frame Start Detection**

* **Schmidl–Cox metric** with training symbol of two identical halves:

 ⁣M[d]=∣P[d]∣2(R[d])2,  P[d]=∑n=0N/2−1rd+nrd+n+N/2∗\!M[d]=\frac{|P[d]|^{2}}{(R[d])^{2}},\; P[d]=\sum\_{n=0}^{N/2-1} r\_{d+n}r^\ast\_{d+n+N/2}

Peak at correct offset gives sample synchronisation.

**5.2 Carrier-Frequency Offset (CFO)**

Estimate fractional CFO

ε^=12π(N/2)arg⁡P[d]\hat{\varepsilon}=\frac{1}{2\pi(N/2)}\arg P[d]

and compensate via complex exponential rotation before FFT.

**5.3 Clock-Drift & Sample-Rate Offset**

The laptop’s DAC/ADC clocks differ by ±100 ppm. Track residual frequency offset using pilot-phase tracking:  
Δf=12πTsym[arg⁡Yk(i)−arg⁡Yk(i−1)]\Delta f = \frac{1}{2\pi T\_{\text{sym}}}\bigl[\arg Y\_k^{(i)} - \arg Y\_k^{(i-1)}\bigr].

**5.4 Practical Hints**

* Window correlation to first 0.5 s to avoid false peaks from echoes.
* Use a 128-sample CP >L>L, but exclude CP from metric to reduce CFO bias.

**6 Experimental Design Guidelines**

* **Hardware:** identical laptop model preferred; disable AGC; set speaker volume <80 %.
* **Sampling:** 48 kHz/16-bit PCM WAV throughout project to maintain inter-operability.
* **Room preparation:** measure background noise, apply 20 dB SNR margin; record at least 55 MLS periods.
* **Software chain:** *Python/numpy*, *scipy.signal.fftconvolve*, JSON metadata for impulse response file.
* **Validation:** convolve known test wave-file with estimated h[n]h[n]; run bit-error-rate (BER) after equaliser, target <10−3<10^{-3} at 2 kbit s⁻¹.

**7 Discussion of Trade-offs**

| **Criterion** | **MLS** | **Log-Sweeps** | **OFDM Pilots** |
| --- | --- | --- | --- |
| Measurement time | Short | Long | Embedded |
| Non-linear immunity | Poor | Excellent | N/A |
| Implementation complexity | Low | Moderate | Low |
| Dynamic range | Limited | >100 dB | 30–40 dB |
| Best use-case | Quick lab test | Final calibration | Continuous tracking |

Zero-padding (ZP) instead of CP avoids circular noise but complicates equalisation; gains are marginal in highly reverberant rooms.

**8 Conclusions and Future Work**

A hybrid scheme is recommended: perform an initial high-resolution chirp sweep to design the equaliser; embed sparse pilot-OFDM blocks to track slow channel drift. Future work may explore adaptive filtering and *deep-learning-aided* synchronisers for moving-speaker scenarios.

**9 References**

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*(Additional contemporary sources consulted are embedded via inline citations throughout the text.)*