Errata of "SDR Implementation of Analog FM Broadcast Multipath Filter"

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This errata list is for the following technical report:

Kenji Rikitake, "SDR Implementation of Analog FM Broadcast Multipath Filter", IEICE Technical Report, vol. 121, no. 227, SR2021-43, pp. 17-24, November 2021.

Errata

The following errors were found by the author on 22-OCT-2021. The author has reported the errors to IEICE SR SIG.

- In Page 21, Table 4, the value of NLMS coefficient update rate is shown as: "48kHz (once in 8 samples)". This should be corrected and shown as: "96kHz (once in 4 samples)".
- In Page 21, at the 8th line from the bottom on the left column, a part of the paragraph is shown as "[...] empirically set to 48kHz to [...]". This should be corrected and shown as "[...] empirically set to 96kHz to [...]".

[End of document]

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ソフトウェア無線によるアナログFM放送マルチパスフィルタの実装

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あらまし　マルチパスによる受信障害の低減はアナログ FM 放送の高品質受信にとって不可欠である．過去に適応フィルタによる解決法が示されているが，その実装詳細は明らかではない．本報告では，筆者が開発した IF 出力に対し複素 NLMS フィルタを適用したオープンソースの SDR 実装の詳細について説明し，開発上の留意点とマルチパスフィルタによる改善された受信結果の評価について述べる．実験の結果，提案したマルチパスフィルタは 880Hz の時報音の放送にて THD+N を 1.22% から 0.33% まで低減し，簡易に設置されたアンテナでも効果的に受信品質を上げることを示した．
キーワード　マルチパス，適応フィルタ，FM 放送, SDR

SDR Implementation of Analog FM Broadcast Multipath Filter

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Abstract　Reduction of the reception degradation caused by multipath propagation is essential for the high-quality reception of analog FM broadcast. Previous reports show adaptive filters effectively solve this issue, but the implementation details are not well-described. In this report, the author explains the details of an open-source SDR implementation that applies a complex NLMS filter to the IF output developed by the author. The author also describes the critical points to consider for the development and the evaluation of the improved reception results by the multipath filter. Experiments showed that the proposed multipath filter reduced the THD+N of an 880Hz time tone signal broadcast from 1.22% to 0.33% and that the multipath filter effectively improved the reception quality for simple antenna installations.
Key words　Multipath, adaptive filter, FM broadcast, SDR

1 Introduction

Quality degradation of radio communication's frequency modulation (FM) scheme has been observed since the beginning of the communication experiments after Armstrong [1] invented the modulation scheme in 1933. Corrington [2] published one of the first mathematical analyses of FM multipath in 1945. Corrington concluded that the drops of carrier modulation amplitudes caused high-level output audio distortion, resulting in spikes in the output audio waveform, which causes annoying sounds and degrading the quality.

Solving the distortion issue of FM multipath is challenging due to the nature of multipath propagation, especially in the urban environment where the reflection of airwaves caused by many buildings exists, which causes a complex pattern of frequency-selective fading. Using directive antennas for solving the distortion issue is not necessarily sufficient to reduce the unwanted reflected waves. It is not practical either because of the large size and mechanical fragility of the beam antennas, particularly for smaller antenna installations such as those on balconies of an apartment complex and on mobile vehicles.

Analog FM broadcast stations in Japan have been a popular source of quality audio contents since 1969 when the national broadcaster NHK-FM [3] and FM Tokai (now TOKYO FM [4]) started the official broadcasting operation. FM broadcast is more resistant to urban electrical noise than amplitude modulation (AM) broadcast. Commercial broadcasters in Japan have announced in July 2021 that they will start replacing the AM radio stations with the equivalent FM stations by fall 2028 [5] to reduce service costs and improve reception sound quality.

Degradation of the transmission quality caused by the multipath propagation has been a persistent issue of FM broadcast reception. Removing the effect of received reflected signals cannot be done trivially by adding a received signal amplifier to the antenna cable or adding a filter to the demodulated audio signals.

Eliminating the root cause of the degradation caused by the interference of receiving multiply delayed and echoed signals by the multipath propagation requires a finite impulse response (FIR) filter for the adaptive cancellation. However, this adaptive filter function is not popular among commercially available FM receivers, and the known receivers with this function are expensive and not generally available for the consumers[1].

On the other hand, inexpensive software-defined receivers (SDRs) become more available at a lower price as the technology matures. In addition, the signal processing performance of commercially available computers has also been increasing rapidly and can handle intermediate frequency (IF) signals from the SDR frontends directly under the order of mega samples per second. Regarding the availability of more computing and processing powers, implementing an FM multipath filter for a receiver with an SDR frontend and a computer by the adaptive cancellation of the interfering signals becomes a practical solution for improving the general reception quality of the FM broadcast signals.

In this paper, the author describes the design and implementation details of an FM multipath filter on the author's SDR software called *airspy-fmradion* [8] and evaluates how the filter has improved the reception signal quality of FM broadcast stations. In Section 2, the author introduces and explains the related work. In Section 3, the author describes the design details of the FM multipath filter of *airspy-fmradion*. In Section 4, the author shows the performance evaluation of the implemented filter. Finally, in Section 5, the author concludes the paper and shows the issues for future work.

2 Related Work

ITU-R presents the overall specification that defines the FM stereo broadcast signal as the Recommendation BS.450 [9] Section 2.2 (Pilot-Tone System). This specification was initially an approved standard of US FCC in April 1961 [10] and has been the national standard in Japan since July 1968 [11].

The relationship between time delays caused by the multipath propagation and the result of demodulated FM signals has been analyzed both in analytic and numerical manners. Engel [12] proposed a model of demodulated signals from the multipath structure of the propagation channel in 1969. Ohara [13] showed an analysis of the desired-to-undesired channel ratio (D/U ratio) and the multipath distortion and how the distortion significantly affected the subjective listening quality of the stereophonic signal by a computer simulation in 1980. Kondo and Date [14] detailed simulation results of applying a transversal filter (equivalent to FIR filter in the discrete digital domain) for erasing the distortion caused by the echoes (equivalent to multipath signals) in the received signal of FM broadcast in 1983.

Godard [15] proposed the methodology of the Constant Modulus Algorithm (CMA) in 1980. Treichler and Agee [16] showed a detailed analysis of CMA by simulation in 1983. Trichler and Larimore [17] later extended the CMA for faster processing with non-complex real signals and known envelope characteristics in 1985.

CMA applies the Least-Mean Square (LMS) [18] algorithm for adaptive filters to the complex number representation [19] of the in-phase and quadrature (IQ) signals. The goal of CMA is to filter out the delayed multipath signals by applying the adaptive FIR filter so that the amplitude of the signal is closest to the given constant value, under the assumption that the amplitude of the transmitted FM signal is a constant and that the amplitude of the received signal changes solely due to the effects of the propagation environment. Mochizuki and Hatori [20] showed detailed simulation results of applying a complex LMS filter and the CMA for the IQ signal to received FM broadcast signals in 1985. Kammeyer [21] shows an optimization method of the complex equalizer coefficients of the CMA in 1986. Hayashi [22] announced an FPGA-based FM stereo receiver prototype production in 2018, including a similar multipath filter algorithm with CMA.

There are other methods to reduce the multipath-derived distortion of the received FM broadcast signals than CMA. Ishigaki, Muraoka, and Hagiwara [23] presented a multipath reduction system based on a pre-calculated model of a single reflected wave with analog processing circuits in 1980. Otani, Tong, and Akaiwa [24] proposed a multiple antenna diversity FM broadcast reception system to minimize the distortion power of the stereo pilot signal. Moir and Pettigrew [25] showed a signal regeneration method with their proposed amplitude-locked loop in analog and SDR circuits without adaptive filters specifically for mobile reception environments in 2014. Chen, Baker, McCarthy, and Thyssen [26] proposed a noise estimation and audio noise reduction method based on the observed L-R stereo subcarrier and the modulated signals in 2015.

3 FM Multipath Filter Design for SDR *airspy-fmradion*

In this section, the author explains the design details of the SDR *airspy-fmradion* and the FM multipath filter.

3.1 Overview of *airspy-fmradion*

The SDR *airspy-fmradion* is a forked software from two previous SDR open source software packages called SoftFM [27] and NGSoftFM [28]. SoftFM implemented an FM stereo SDR receiver with the SDR frontend support for RTL-SDR [29], [30] and the basic support of stereo signal decoding of the Pilot-Tone System. NGSoftFM enhanced the SoftFM with support for various frontends such as AirSpy R2/Mini [31], [32].

Figure 1 shows the architectural diagram of *airspy-fmradion*. The SDR software processes the IQ signal input from either the SDR fron-

[1] According to the author's survey as of September 2021, only one consumer receiver [6] and one professional monitor receiver [7] have the function of FM multipath filtering. The implementation details of the filters are unknown because they are proprietary products.

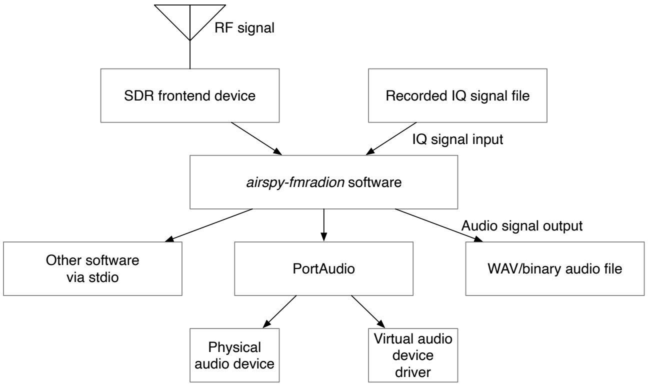

Fig. 1 *airspy-fmradion* functional diagram.

Table 1 *airspy-fmradion* specification summary.

Supported SDR frontends: Airspy R2/Mini, Airspy HF+ Dual Port [33]/Discovery [34], RTL-SDR, and reading recorded IQ signals from WAV and binary files
Supported operating systems: macOS, Linux (Ubuntu/Debian), and Raspberry Pi OS
Downsampled IF bandwidth from SDR frontends: 384kHz for FM broadcast [2], 192kHz for the other modes
Received output formats: 48kHz mono/stereo audio data in 16-bit signed integer or 32-bit floating-point values
Output destinations: binary raw data stream to the standard output; files in WAV/RF64 [35] and the raw binary format, with the optional timestamp text file; and OS audio output devices including physical audio devices and virtual device drivers through PortAudio [36] library

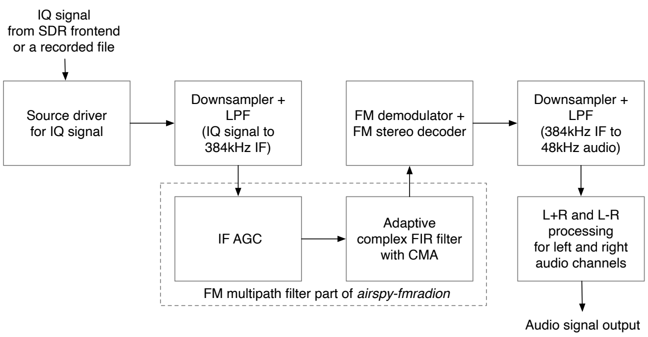

Fig. 2 *airspy-fmradion* FM broadcast receiver diagram.

tend devices or the recorded IQ signal file and outputs the demodulated audio for the multiple output objects.

Table 1 shows the specification summary of *airspy-fmradion* [37]. This paper focuses on describing the FM broadcast functions of *airspy-fmradion* and the experimentation results.

Figure 2 shows the software components within *airspy-fmradion* when the software processes FM broadcast signal. The source driver is specific for each frontend device, including the IQ signal file reader. The output sample rate of the source driver differs due to the limitation

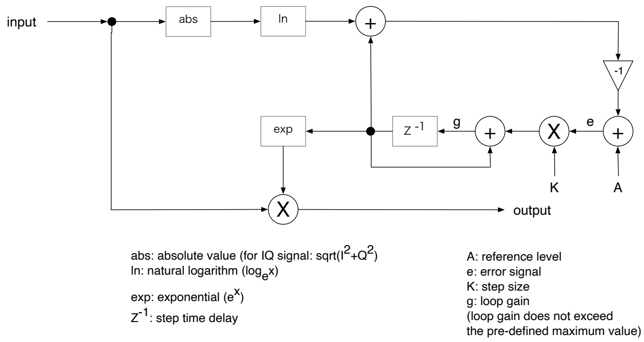

Fig. 3 IF AGC block diagram for *airspy-fmradion* FM multipath filter.

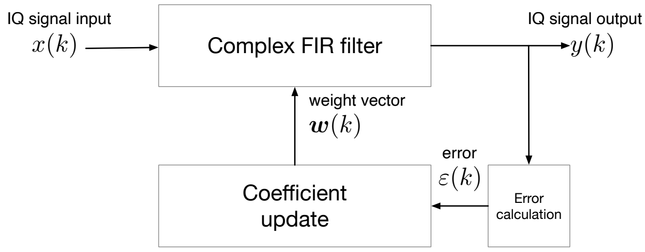

Fig. 4 Adaptive complex FIR filter block diagram for *airspy-fmradion* FM multipath filter.

of the frontend [3], so the output sampling rate has to be normalized to the IF sampling rate of 384kHz by a fractional downsampler of the libsoxr [38] library and be optionally applied the IQ low-pass filter (LPF) for a narrower signal bandwidth to reduce interference.

3.2 FM Multipath Filter Design Details

In this section, the author explains the FM multipath filter part shown in Figure 2 and the design details. The multipath filter part is split into two blocks: the IF automatic gain control (AGC) and the adaptive complex FIR filter with CMA.

Figure 3 shows the IF AGC algorithm [4] prepended before the FM broadcast multipath filter [5]. This AGC algorithm, called logarithmic-loop AGC, aims to minimize the difference between the complex amplitude of the input IQ signal to the reference level A [6], which is suitable for further processing by the multipath filter. The logarithmic-loop AGC algorithm performs well for both AM and FM, which preserves the input amplitude change necessary for later processing in the multipath filter.

Figure 4 shows the adaptive complex FIR filter block for the FM broadcast multipath filter. This filter has the coefficient vector of

[3] For example, the valid sampling rate of Airspy R2 is either 10MHz or 2.5MHz, and for RTL-SDR, it is above 900kHz and below 3.25MHz. The downsampling ratio is not necessarily to be an integer, so a fractional downsampler is mandatory.

[4] The author modified the algorithm from the original one [39] in the following manners: the complex amplitude function (of absolute value) replaced the AGC detector; the logarithmic and exponential functions were moved out of the feedback loop, and an addition replaced a multiplication in the original algorithm for faster computation.

[5] The author could not find any previous example of explicitly prepending an AGC before the FIR filter in the surveyed documents.

[6] A is set to 1.0 for simplifying the later multipath filter design.

length N, the same as the number of FIR filter coefficients. The filter output $y(k)$ at the discrete timing value k changes as the filter input $x(k)$ changes to minimize the error $\varepsilon(k)$ so that the $y(k)$'s complex amplitude is closest to unity (1.0), as configured in the prepended AGC stage.

The following explains the complex LMS algorithm [19] [17, Section II]. Given the discrete time complex input $x(k)$ at time k, the FIR filter output $y(k)$ can be described as

$$y(k) = \boldsymbol{x}(k)^\top \boldsymbol{w}(k) \tag{1}$$

where $\boldsymbol{x}(k)$ is the size-N complex vector of FIR filter input

$$\boldsymbol{x}(k) = [x(k), x(k-1), \ldots, x(k-N+1)]^\top \tag{2}$$

and $\boldsymbol{w}(k)$, the size-N complex vector of FIR filter coefficient, is given by

$$\boldsymbol{w}(k) = [w_0(k), w_1(k), \ldots, w_{N-1}(k)]^\top \tag{3}$$

where the index k shows that the coeffients are adjustable in time.

The complex error value $\varepsilon(k)$ to the reference amplitude level of unity is given by

$$\varepsilon(k) = \{\|y(k)\|^2 - 1\} \cdot y(k) \tag{4}$$

where $\|y(k)\|$ is L^2 norm of $y(k)$.

$\boldsymbol{w}(k+1)$, the updated coefficient vector for time $k+1$ from $\boldsymbol{w}(k)$ is given by

$$\boldsymbol{w}(k+1) = \boldsymbol{w}(k) - \mu\varepsilon(k)\boldsymbol{x}^*(k) \tag{5}$$

where each element of $\boldsymbol{x}^*(k)$ is the complex conjugate of the respective element of $\boldsymbol{x}(k)$ and μ is the adaptation gain or step size, which determines the convergence characteristics.

In LMS, choice of the fixed adaptation gain value μ, which is also critical for the stability of the algorithm, is not trivial and the optimal value is difficult to determine. The author decided to use the Normalized LMS (NLMS) algorithm [40], a variable-step LMS (VS-LMS) algorithm reflecting the input data amplitude, to maximize the stability and faster convergence. Bismor, Czyz, and Ogonowski [41] showed NLMS had an unrivaled advantage for real-time nonstationary data processing by comparing 17 VS-LMS algorithms including NLMS.

In NLMS, μ for time k is calculated dynamically with the vector $\boldsymbol{x}(k)$ and is given by

$$\mu = \alpha/\|\boldsymbol{x}(k)\|^2 = \alpha/(\boldsymbol{x}(k)^\top \boldsymbol{x}^*(k)) \tag{6}$$

where α is a real number constant to determine the adaptation gain. Slock [42] proved NLMS converges when $0 < \alpha < 2$. From the equations (5) and (6), the coefficient update algorithm of complex NLMS is given by

$$\boldsymbol{w}(k+1) = \boldsymbol{w}(k) - \alpha\varepsilon(k)\frac{\boldsymbol{x}^*(k)}{\|\boldsymbol{x}(k)\|^2} \tag{7}$$

Table 2 Reception conditions for *airspy-fmradion* FM multipath filter.

Location	35.67N, 139.63E (accuracy: ±0.01 degrees)
Antenna	Diamond Antenna AZ510FMH (91cm-length whip antenna), mounted horizontally at 8m-high above the ground with 5m×5 counterpoise wires, directed to 15 degrees east from the true north (antenna directivity: approx. east-west), with 15m 5D-FB coaxial cable, 1:2 splitter, and 20cm RG-316/u coaxial cable
Receivers	Airspy HF+ Discovery and RTL-SDR V3
Software	*airspy-fmradion* Version 20210817-0, compiled with gcc 10.3.0, and libvolk [43] Version 2.4
Computer	Intel NUC NUC10i7FNH with Intel Core i7-10710U CPU, 16GB RAM, Linux 5.11.0-31-generic, Ubuntu 21.04 x86_64

where $\|\boldsymbol{x}(k)\|$ is the L^2 norm of $\boldsymbol{x}(k)$.

Note well that the frequency of coefficient update of LMS/NLMS algorithms does not have to be every time the filter calculation is done (i.e., in the filter sample rate), regarding the slower time variance of the radio wave propagation environment.

4 Evaluation of the FM Multipath Filter

In this section, the author shows the experimentation results of the *airspy-fmradion* FM multipath filter and the evaluation.

4.1 Measurement Indices

The author introduces two methods for measuring reception quality. The first one is total harmonic distortion plus noise (THD+N) [44], [45]. THD+N is suitable for measuring multipath-induced distortion of a pure sine-wave transmission, which happens regularly and periodically on NHK-FM time tone transmission of 440Hz and 880Hz. Research of aural detection thresholds [46] suggests THD of 0.6% is detectable and affects the subjective evaluation of listening audio quality.

The second one is Quadrature Multipath Monitor (QMM) [47]. Decoding the L-R audio signal from the received composite audio requires multiplying the regenerated 38kHz clock from the received 19kHz pilot signal with the zero phase shift [9, Section 2.2.2.5]. On the other hand, multiplication of the $\pi/2$-shifted regenerated 38kHz clock to the received composite audio signal should not generate any output in an ideal propagation condition. However, various distortion and interference results can be observed from the result of this multiplication. QMM employs this multiplication result as the index. QMM is applicable for measuring generic broadcasting content quality, provided that the L-R signal power is significant, such as that of music and commercial advertisements.

4.2 Experiment Setup and Conditions

Table 2 shows the experiment setup [7]. The author decided to use a simple whip antenna to reproduce the condition of casual reception without the beam antennas. The computer in the table had adequate performance in processing the incoming signal without dropping the data from the SDR frontends.

[7] The exact reception location is hidden to protect the author's privacy.

Table 3 FM broadcast station antenna information.

NHK-FM Tokyo (JOAK-FM 82.5MHz) [48]
At Tokyo Skytree, 35.710139N, 139.810833E (~ 17km east of the reception location); Antenna ground height: 540m; Output: 7kW, ERP: 57kW
InterFM Tokyo (JODW-FM 89.7MHz) [49]
At Tokyo Tower, 35.658611N, 139.745556E (~ 11km east of the reception location); Antenna ground height: 320m; Output: 10kW, ERP: 13kW

Table 4 Parameters of *airspy-fmradion* FM multipath filter.

Overall sampling rate		384kHz (~ 2.6μs/sample)
IF AGC	initial gain	1.0
	maximum gain	10000.0 (80dB)
	reference level A	1.0
	step size K	0.001
NLMS	adaptation gain α	0.1
	coefficient update rate	48kHz (once in 8 samples)
FIR filter	$\boldsymbol{w}(k)$ initial values ($j^2 = -1$)	$1 + 0j$ at the reference position, $0 + 0j$ for all the other elements
$\boldsymbol{w}(k)$ value conditions: for S-stage filter, the reference position is $k = 3S+1$, with $3S + 1$ elements of the past samples and $S - 1$ elements of the future samples; the filter order $N = 4S + 1$; and the imaginary part value is always 0 for the coefficient at the reference position element $w_{3S+1}(k)$.		

Table 3 shows the FM broadcast station antenna locations and details used for the experimentation. The distance between the reception location and the antennas was less than 20km. The direction was approximately at the east, within the directivity range of the antenna for reception, and many reflected radio waves from the local buildings were expected. The signal strength measured by Airspy HF+ Discovery for NHK-FM Tokyo and InterFM Tokyo were $\simeq -24$dBFS [8], which was adequate for the multipath filter evaluation.

Table 4 shows the parameters of the FM multipath filter. The IF AGC configuration retained the amplitude variance information necessary for the multipath filtering. The author empirically set the NLMS adaptation gain to 0.1 for stability. This gain value was much higher than the fixed gain value $\mu = 10^{-5}$ in [20], but the filter remained stable during the real-world signal processing of NHK-FM and InterFM radio waves at the reception location. Results of the reception with a 15-stage filter showed the QMM output of the filter converged within 0.3 seconds from the beginning of the reception. The coefficient update rate was also empirically set to 48kHz to reduce the processing load while maximizing the noise reduction.

The reference point in the initial coefficient vector is where the element value is first set to unity and whose imaginary part is always reset to zero during the filter processing for the filter stability. The element is intentionally positioned to reflect the expertise that the past samples affected more than the future ones. The author recorded the IQ output signal from the SDR frontends and experimented by decoding the played-back signal data in different conditions, such as changing the number of filter stages.

[8] The unit dBFS stands for decibels relative to full scale (0dBFS) [50].

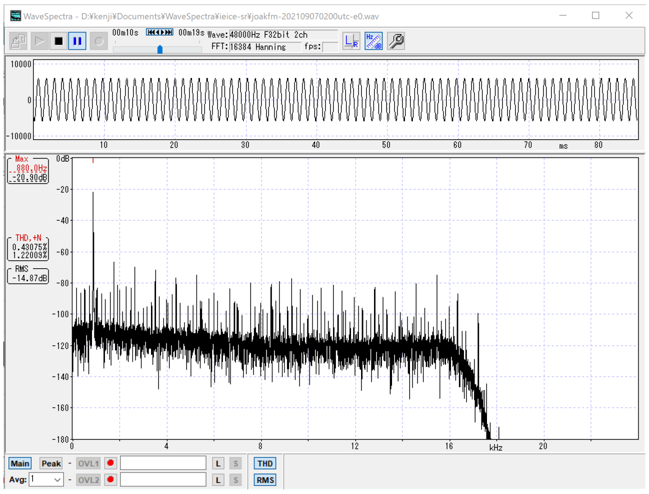

Fig. 5 NHK-FM Tokyo time tone spectrum without the multipath filter.

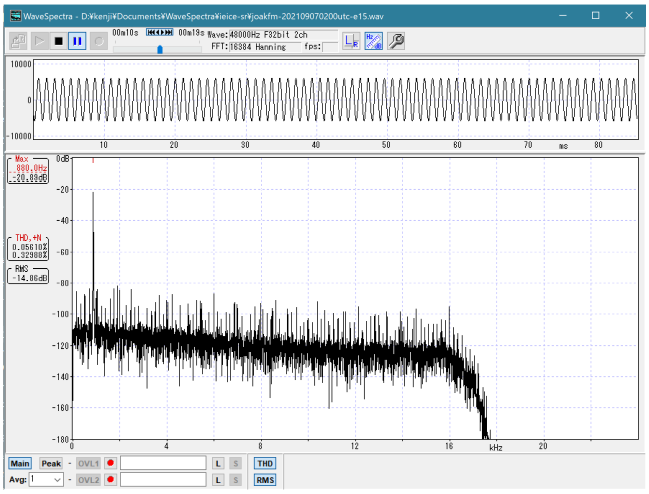

Fig. 6 NHK-FM Tokyo time tone spectrum with the 15-stage multipath filter.

4.3 Experiment Results

Figure 5 shows the received audio spectrum of an 880Hz time tone output broadcast by NHK-FM Tokyo at 11:00 Japan Time (02:00 UTC), September 7, 2021, by WaveSpectra [51]. Figure 6 shows the audio spectrum after the 15-stage multipath filter was applied. The harmonics of 880Hz tones were reduced by the filter more than 10dB.

The author experimented by incrementally changing the number of filter stages from 0 to 10 by one, and 15, 20, 30, 40, and 50. Figures 7, 8, and 9 show how the THD+N level [9] and the QMM output of the 880Hz time tone, and the noise level of no-sound output in the time tone announcement changed by the number of filter stages. The QMM level decreased maximum in 10.6dB, and the no-sound noise level decreased maximum in 2.26dB. The similarity of THD+N and

[9] THD+N was measured by WaveSpectra [51] 1.51, for 48kHz 32bit floating-point stereo format, with the FFT of 16384 samples and the Hanning window, left channel, at the point of 480000 samples (10 seconds) from the beginning.

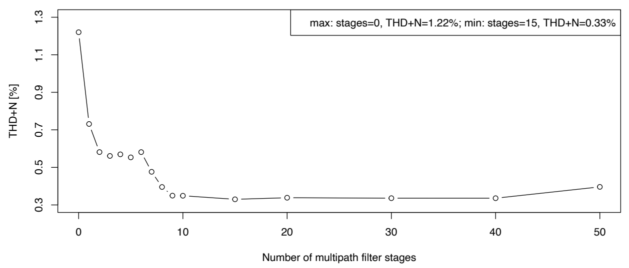

Fig. 7　THD+N of NHK-FM Tokyo time tone.

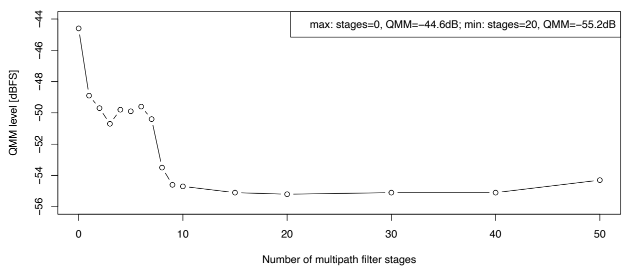

Fig. 8　QMM output of NHK-FM Tokyo time tone.

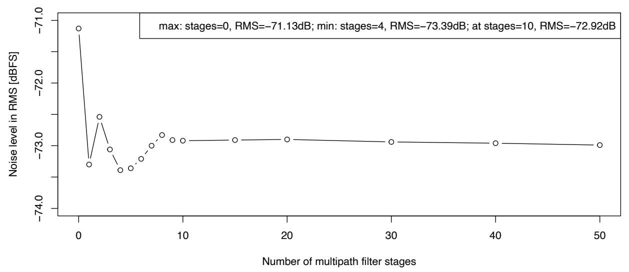

Fig. 9　RMS level of NHK-FM Tokyo no-sound output.

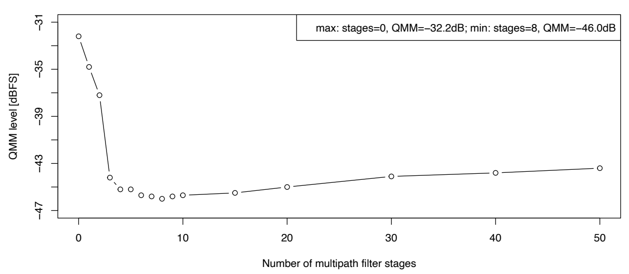

Fig. 10　QMM output of InterFM Tokyo received by Airspy HF+ Discovery.

QMM level characteristics suggests that either one could be used as an index for distortion measurement of the received audio signal.

The author also experimented with how the QMM output level changed by changing the number of filter stages for InterFM Tokyo sig-

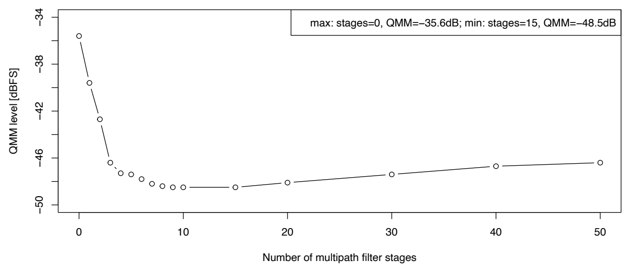

Fig. 11　QMM output of InterFM Tokyo received by RTL-SDR.

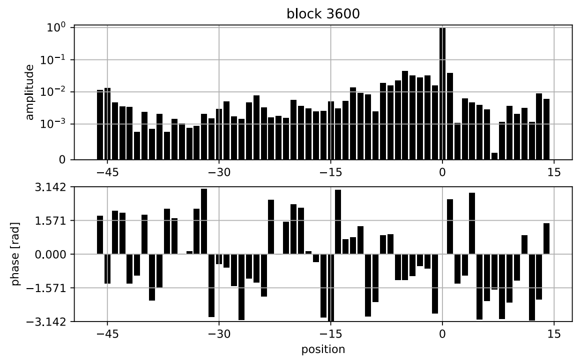

Fig. 12　Coefficients of the FIR filter for NHK-FM Tokyo reception.

nals received by two different SDR frontends at two different sources. Figure 10 shows the result of the QMM level after 20-second reception from 16:13:12 Japan Time (07:13:12 UTC) on September 7, 2021, by Airspy HF+ Discovery. Figure 11 shows RTL-SDR's 20-second reception from 9:48 Japan Time (00:48 UTC) on September 13, 2021. The contents of both receptions were pop music. In either case, the QMM level decreased maximum in 13 ∼ 14dB. This result suggests that the proposed filter also worked well on the musical contents.

Figure 12 shows the coefficient vector in polar form for the NHK-FM Tokyo time tone reception after 3600 blocks (∼ 0.96 seconds) of reception, showing the estimated delay profile of the multipath propagation environment. Excluding the signal at the reference point, the amplitude of the past samples was exponentially decreasing, which fits well with the Rayleigh fading model [52]. Furthermore, the amplitude of the future samples from the reference point is approximately 1/10th of those of the past samples, which suggests that allocating a shorter vector for future samples does not hamper the filter performance. On the other hand, the higher amplitude level of both past and future ends in the coefficients suggests that a larger vector size is required for more accurate delay profiling of the multipath propagation.

4.4 Observations

The author measured the processing performance difference on the computer shown in Table 2 with how the single-instruction multiple-data (SIMD) parallelization library libvolk [43] was configured and optimized for each computer. The CPU usage with 100 stages of the FM multipath filter given the optimal configuration for libvolk was ∼ 19% of a single core when the `scaling_governor` [53] was `performance` mode and the `scaling_max_freq` was 4700000 (4.7GHz). On the other hand, the CPU usage was ∼ 43% without the SIMD optimization, when all libvolk functions were configured to use generic functions written in C. This observation suggests that floating-point vector calculation still consists of significant CPU usage and that optimization is crucial [10]. The author successfully executed *airspy-fmradion* on a Raspberry Pi 4 Model B [55] with enabling the multipath filter by the libvolk optimization.

The author also observed that the prepended IF AGC made the multipath filter more robust in receiving weaker station signals by keeping the signal amplitude closer to the target value at the FIR filter input. The author learned the effect while listening to radio stations in Yokohama [11], which had weaker signals than those in Tokyo and whose line-of-sight radio waves were blocked behind large buildings.

CMA requires the assumption that the propagation channel preserves the amplitude change of modulated signals. This assumption is false for some propagation channels. For example, the FM broadcast signals relayed through the cable TV links are through a hard limiter which sets the signal amplitude to a fixed value and removes the information in the amplitude change. The author observed that the proposed multipath filter in this paper was not effective for the signals received through the cable TV links.

While QMM is effective as a quality index of highly modulated signals such as pop music, some contents such as classical music and news reading contain low-level modulated signals and do not necessarily contain rich L-R components. Under such circumstances, another measurement index of multipath distortion is necessary. Measuring the THD+N of the 19kHz pilot tone of the FM broadcast [24] after filtering the tone signal to remove other modulation signals is a possible candidate, though handing the signal delay between the demodulated pilot tone and the error calculation of the FIR filter to minimize the distortion will be a hard-to-solve problem.

5 Conclusions

This paper described the historical significance of reducing multipath distortion for FM broadcast reception and presented the theoretical background, design details, and evaluation of the FM multipath filter of the author's open-source SDR receiver *airspy-fmradion*.

The experiment of the proposed multipath filter with the real-world FM broadcast stations, not by the simulated signals such as those in the previous research papers, showed that the filter significantly reduced the multipath distortion to improve the objective audio quality indices such as THD+N and QMM output level. The proposed multipath filter could reduce the THD+N from above 1% to below 0.6%, which proved that the filter was a practical solution for FM broadcast listeners with simple antennas to improve their listening quality. The author also showed that implementing the filter on modern computers was feasible by applying CPU usage optimization and the empirical parameter choice.

On the other hand, the proposed multipath filter and the quality measurement indices do not cover all the possible conditions of FM broadcast reception, such as that via the cable TV links. For such cases, alternative quality indices and filtering methods such as measuring the distortion of the 19kHz stereo pilot-tone signal and setting the adaptive filter to minimize the distortion should be considered.

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