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Improvement of Sound Quality in FM Stereo with Monitored Audio Processor Technology

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It is known from published papers that the coaxial cable and antenna system of an FM stereo station often reduce stereo channel separation, deteriorating the quality of the emitted sound. This problem is motivated by impedance mismatch errors caused by the characteristics of the radiating dipoles. In this work, a mathematical analysis of this problem will be performed using SPICE simulation to identify its origins. Based on the results of this analysis, the solution that has been developed based on the design of a monitored audio processor will be presented. This technology allows achieving high channel separation using the existing antenna and adds the capability to remotely measure, adjust, and certify the quality of the emitted sound.

0 INTRODUCTION

The FM stereo broadcasts began in 1961, marking the progress of a technology that became synonymous of high sound quality, surpassing the magnetic recordings of that era.

With the advent of digital radio technologies such as digital radio Mondiale (DRM) or digital audio broadcasting (DAB) [1] in this century, a rapid conversion of FM radios to these new systems was expected because of the promised sound quality. However, the experience proved otherwise.

The limited growth of digital radio was due to the disappointment of listeners who did not find any audible improvement in sound quality. This is because digital broadcasts require the use of perceptual encoding systems such as MP3, AAC, etc., which, although a significant advancement in streaming technology [2], are a limitation in the sound quality that broadcasting transmissions require. Additionally, perceptual encoding does not tolerate double encoding, which forces FM stations to use music that has never been previously encoded, which is difficult to achieve. Numerous engineers have documented these limitations [3].

In contrast, FM stereo does not require any type of perceptual encoding and currently uses FM transmitters with digital front ends and direct digital synthesis modulation. This reduces distortion to inaudible levels and achieves dynamic ranges of 85 dB or even better.

An FM radio station that uses a digital internet protocol (IP) console connected via USB to a PC that plays audio can digitally input to the transmitter via AES3 [4]. This maintains a signal quality of 24 bits/48 kHz from the audio file to the radiated signal. The advantage of FM without any perceptual encoding is evident. Its on-air quality is always excellent because musical pieces can be stored in nonencoded pulse code modulation (PCM) WAV files or in high-bit-rate MP3 or Opus files. For this reason, analog FM broadcasts remain the primary method of sound broadcasting worldwide, and engineers continue to make efforts to improve its quality in both transmission and reception [5].

Radio stations currently operate 24 hours a day because of the invention of programming automation systems in 1988 [6], which means a significant portion of their broadcast consists of music, demanding the best possible sound quality.

Since the beginnings of FM stereo, the Federal Communications Commission (FCC) regulations in the United States have set a minimum channel separation value of 30 dB [7]. This value has been adopted in numerous countries, whereas others, such as Japan, require 33 dB [8]. Today, after 60 years, broadcast engineers design installations to achieve a minimum of 35 dB to 40 dB of channel separation, as the industry offers stereo encoders that exceed 60 dB of separation [9]. However, in numerous measurements conducted on the emitted air signal, it was found that this value of 35 dB is rarely achieved in FM stations, especially those with less than 5 kW of power, due to the use of serially manufactured antennas that are not individually adjusted on the transmitting tower along with their power splitters.

This phenomenon of reduced stereo separation is not new, as it was noticed in 1966 by Onnigian [10], who studied a real antenna with different levels of impedance mismatch that generated standing waves (voltage standing wave ratio [VSWR]). This led to the conclusion that the increase in standing waves generated in the coaxial cable due to the impedance mismatch of the antenna is the cause of reducing stereo separation values to less than 20 dB, which implies a severe deterioration in sound quality and noncompliance with transmission standards. This common problem, which until now had no solution, nullifies the efforts made by radio stations to improve sound quality through digital equipment. Producers of radio programs often complain that despite using expensive audio processors in their FM stations, they fail to achieve the same sound quality on FM receivers as they hear in the control room monitors.

Therefore, the first step is to mathematically study the influence of impedance mismatches in the transmitter system, coaxial cable, power dividers, and antenna dipoles in order to correct them and meet the values required by international standards for stereo channel separation.

1. SPICE SIMULATION

Because Onnigian's work is experimental in nature and the relationship between VSWR and channel separation is not clear, it was decided to analyze the origin of this problem. It was noticed that investigating it analytically would be complex because FM signals require the use of Bessel functions for analysis. Although in specific cases, such as the energy transmitted in sidebands, it is possible to find an analytical solution as in Bonello [11], the solution becomes very complex in the case of transmission through a coaxial cable terminated in a reactive load with varying impedance. Therefore, it was decided to approach it as a SPICE simulation of the system: transmitter + coaxial cable + antenna + FM receiver, using the Tina calculation software.

To perform the SPICE simulation, start with the equation that gives the channel separation of a stereo encoder [12]:

$$\mathrm{Sch} = 10\log\frac{(\cos\theta + \frac{S}{M}\cos\varphi)^2 + (\sin\theta)^2}{(\cos\theta - \frac{S}{M}\cos\varphi)^2 + (\sin\theta)^2}, \qquad (1)$$

where:

Sch = Stereo channel separation in decibels
θ = Phase difference between L+R and L-R channels
φ = Phase difference at 38 kHz due to pilot error at 19 kHz
M = Main channel gain (L+R)
S = Stereo channel gain (L-R)

To analyze the problem, the gain errors and phase errors will be considered separately. It was noticed that in both cases, $\varphi = 0$ because the manufacturer of the stereo encoder adjusts this value in each unit to achieve high channel separation. Additionally, assume $\theta = 0$ to isolate the errors caused solely by the gains of M and S. Applying these conditions to Eq. (1),

$$\mathrm{Sch\,(Gain)} = 10\log\frac{(1 + \frac{S}{M})^2}{(1 - \frac{S}{M})^2} = 20\log\frac{M+S}{M-S} \qquad (2)$$

Similarly, to examine the influence of phase rotation on the L-R channel assuming no amplitude variation (M = S), the following is obtained from Eq. (1) with the condition: $\varphi = 0$ and $\frac{S}{M} = 1$

$$\mathrm{Sch\,(Phase)} = 10\log\frac{(\cos\theta + 1)^2 + (\sin\theta)^2}{(\cos\theta - 1)^2 + (\sin\theta)^2} \qquad (3)$$

Performing calculations with Eq. (3) for values of θ less than 15° as is typical in real systems, it was noticed that the terms $(\sin\theta)^2$ in the numerator and $(\cos\theta - 1)^2$ in the denominator can be eliminated as they produce errors smaller than a hundredth of a decibel. Therefore, Eq. (3) can be simplified as follows:

$$\mathrm{Sch\,(Phase)} \cong 20\log\frac{\cos\theta + 1}{\sin\theta} \qquad (4)$$

Eq. (2) gives us the stereo channel separation due to gain errors between the L+R and L-R channels, and Eq. (4) gives us the separation due to phase errors.

The circuit used in the simulation is shown in Fig. 1. Note that the FM transmitter has been replaced by an ideal transmitter represented by a voltage-controlled oscillator (VCO) modulating a carrier at 10.7 MHz with a deviation of 75 kHz. This frequency has been chosen to simplify the circuit because it corresponds to the intermediate frequency (IF) of the receiver and the frequency discriminator, eliminating the heterodyne system of the receiver. In other words, this work has performed a frequency shifting simulating a real transmission at 107 MHz, within the FM band of 88–108 MHz.

Therefore, the Transmitter-Coaxial-Antenna circuit has been scaled by a factor of 10. Now, it is necessary to simulate a lossless 100-m cable at 107 MHz; this is the typical cable length used to feed a 90-m mast antenna. These 100 m at 107 MHz are equivalent to 1,000 m scaled to 10.7 MHz. Taking into account that the propagation velocity is 89 % of the speed of light in a typical foam cable, the calculation gives a delay of 3.74 ns per meter of cable.

In other words, it would be a 374-ns delay for 100 m in the 107-MHz FM band. But because we are scaling 10:1 to maintain the same wavelength, 1,000 m of cable will be used, which will be simulated with 10×374 ns = 3.74 µs at 10.7 MHz. This is the value that TL1 will have, which is the SPICE simulation of a coaxial line.

The input of the artificial line is coupled to the transmitter by low impedance that is equivalent to the internal impedance of the transmitter. It is generally an indeterminate value and does not affect the calculations, so 1 Ω is used.

At the output of the coaxial TL1, there is a marked VSWR point where the standing wave ratio that appears in the simulation will be measured. It is measured by applying its definition, that is, as the maximum value of the 10.7-MHz carrier voltage divided by the minimum at the output of the coaxial cable (VSWR point).

The simulations are performed in transient analysis mode (TR mode). For the analysis to be accurate, it is necessary to use very small TR steps, less than 1 % of a picosecond, which makes the simulation very slow in the audio range,

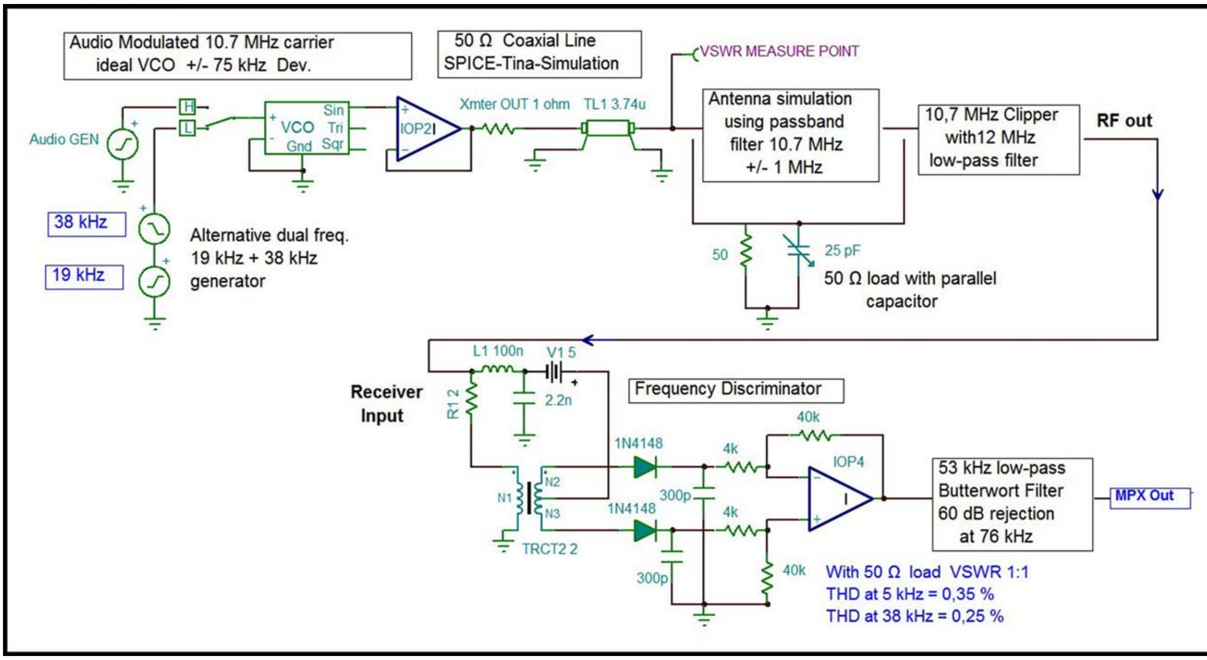

Fig. 1. Block diagram of the SPICE simulation circuit.

Table 1 Stereo channel separation due to gain errors between L+R and L-R channels

Load Impedance	VSWR	Channel Separation [dB]
50 Ω	1:1	40.3 dB
50 Ω // 100 pF	1.38	40.3 dB
60 Ω	1.20	36.5 dB
70 Ω	1.40	36.4 dB
Antenna Fig. 2	1.15	37 dB

requiring long simulation times. Here, another advantage of scaling the carrier frequency is appreciated, because it speeds up the simulation by ten times. To perform the measurements, three types of loads will be used:

A: Resistive load,
B: 50 Ω resistor in parallel with a capacitor or inductor, and
C: Simulated antenna shown in Fig. 2

This study will start by analyzing the channel separation based on the level difference between the L+R channel by applying a 1-kHz tone to the VCO and the L-R channel by applying a 38-kHz tone to the VCO. Then Eq. (2) is used to obtain the channel separation, and the results can be seen in Table 1.

It can be seen that the errors in the output gain at 1 kHz and 38 kHz result in minimal differences in channel separation. Furthermore, it is not related to the standing waves of the antenna and cable, which contradicts the previous belief before this study.

Next, what happens with phase variations will be studied. This is done considering that the 19-kHz pilot carrier generates the 38-kHz signal in the receiver, which replaces the suppressed carrier, and therefore, both will be in the same phase. Therefore, the θ phase errors will be due to the combination of the transmitter + coaxial cable + antenna, and those will be calculated using Eq. (4).

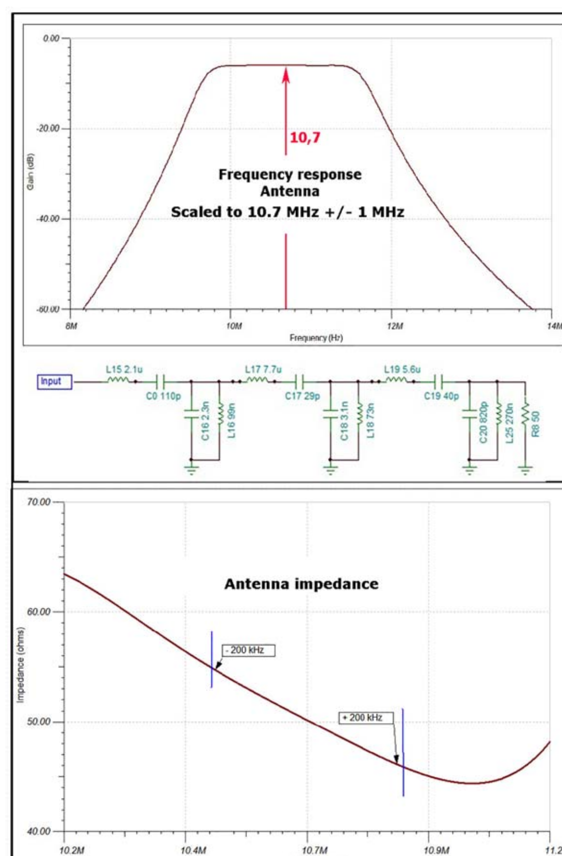

Fig. 2. Passband filter that simulates an ideal antenna

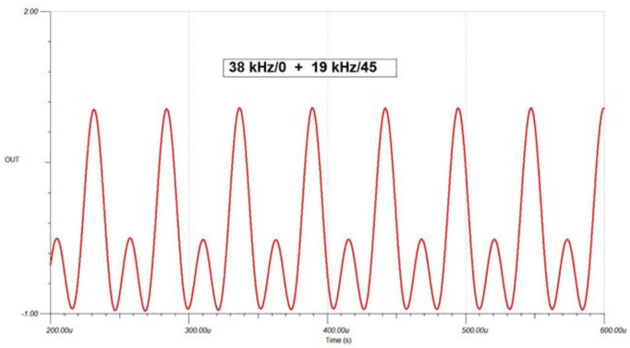

Fig. 3. Sum of 38 kHz/0° + 19 kHz/45°.

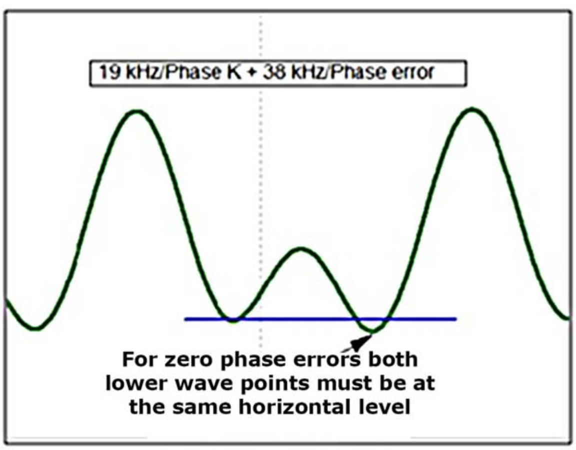

Fig. 4. Dual-frequency wave during the correction process.

This analysis is performed with the same simulation circuit as in Fig. 1, using the dual-frequency generator composed of two sinusoidal oscillators in series, one at 19 kHz and the other at 38 kHz, with equal output voltage. The idea is to measure the phase changes between both frequencies caused by the different types of antenna load. To measure phase differences smaller than 1° through the waveform, a graphical method using an excitation type whose phase is visually recognizable needs to be found.

Because the simulation is performed in transient mode, the simulator itself cannot accurately calculate the phase. To solve this calculation problem, it was found that a very precise method is to shift the phase of the 19-kHz carrier by +45° while keeping the phase of the 38 kHz at zero. This procedure provides a waveform that is easily recognizable at the output of the simulation, as shown in Fig. 3. It can be seen that the two lower peaks remain perfectly horizontal with a 45° phase difference.

This condition can be verified in the simulation by using two cursors placed at both minimum points, which should coincide in their value. The procedure used has been to calibrate without standing waves (with a resistive load of 50 Ω) by adjusting the phase of the 19 kHz to compensate for the phase rotations of the 53-kHz filter in the discriminator (see Fig. 1). This is done as shown in Fig. 4, keeping the 38-kHz generator at zero phase and modifying the phase of the 19-kHz generator to achieve perfect horizontality. This

Table 2 Stereo channel separation due phase errors at L-R channel

Load impedance	VSWR	Channel Separation [dB]
50 Ω	1 : 1	∞
60 Ω	1.20	∞
50 Ω // 25 pF	1.08	35 dB
50 Ω // 50 pF	1.18	29 dB
50 Ω // 100 pF	1.36	23 dB
50 Ω // 150 pF	1.61	20 dB
50 Ω // 4.4 uHy	1.18	35 dB
Antenna Fig. 2	1.15	28 dB

obviously involves running the simulation again for each modification.

Once perfect horizontality is achieved, the phase of the 19 kHz will no longer touched, as this condition implies that of a factory-calibrated receiver

What will be measured subsequently when placing different antenna loads (by correcting the phase of the 38-kHz generator to achieve horizontality) is the value of the phase correction at 38 kHz that needs to compensate for the value created by the coaxial cable + antenna combination, which will be used to calculate the channel separation for that load using Eq. (4). The values obtained due to the phase variation can be seen in Table 2.

From Table 2, some interesting consequences arise. The first one is that standing waves generated by pure resistive loads do not produce phase errors. This means that there is no direct relationship between stereo separation reduction and VSWR as previously thought since 1966. This discrepancy makes the results impossible to compare with those obtained in [10] because, as the simulation shows, a change in resistive load impedance increases VSWR but does not worsen channel separation.

The second conclusion is that there is a strong correlation with parallel reactance, especially if it is capacitive. In the case of inductive reactance with the same impedance, the reduction in separation is much smaller. Table 2 compares a 50-pF capacitor in parallel with a 4.4-nHy inductor, which have the same reactance at 10.7 MHz and produce the same VSWR. However, the capacitor results in a channel separation of 29 dB, whereas the inductor yields a figure of 35 dB. This shows that phase rotation fundamentally deteriorates stereo channel separation. By performing additional measurements for parallel capacitance values and expressing them in reactance values, the curve shown in Fig. 5 is obtained, which expresses the channel separation as a function of capacitive reactance in parallel with a 50 Ω load.

2. THE PROBLEM SOLUTION REAL EXAMPLES

This study has been able to analyze the problem and understand that the existence of phase rotations in the stereo subcarrier is something we will always have because radiating dipoles have reactive impedances that cannot be avoided in serial manufactured antennas. But even if they were to eliminated, there would still be significant amplitude and

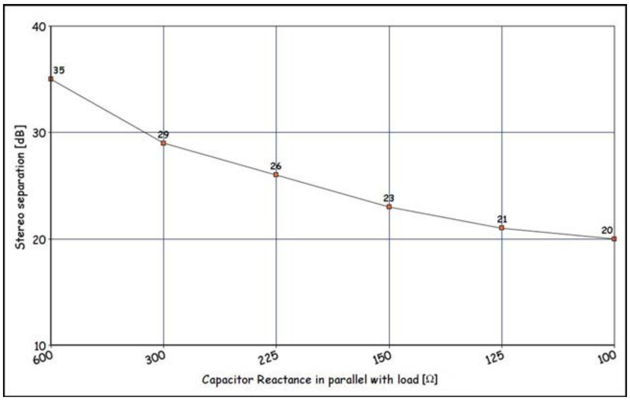

Fig. 5. Stereo channel separation of a 50 Ω load with a capacitor in parallel.

Fig. 6. Digital receiver antenna at the rear side of the audio processor.

phase variations in the studio-transmitter links due to their frequency response.

Despite the efforts made by each manufacturer to improve antennas and links, there will always be small errors in amplitude and phase that reduce stereo separation. That's why the solution found in this study is to correct, prior to the transmitter, the amplitude and phase in the 38 kHz band within the stereo generator. This way, instead of a perfect stereo MPX signal, a corrected signal will be sent to the transmitter, so that after the Studio-Transmitter-Link + Transmitter + Coaxial + Antenna chain, the on-air signal must have a stereo separation greater than 35 dB. In other words, introduce an error in amplitude and phase opposite to what the FM stereo transmission chain produces. This corrected signal will appear at the MPX output to the transmitter or even at the output of the AES3 digital audio used in the new transmitters with internal digital stereo encoder.

It is now evident that the sum of the errors to be corrected is unpredictable and also varies with the aging of antennas, coaxial cables, and equipment. Therefore, they must be measured first and then corrected. In addition, this is not a one-time adjustment due to the aforementioned changes. It is advisable to make periodic readjustments every 6 months, which must be performed without interrupting the transmission.

However, before the creation of monitored processors, there was no way to do these measures without shutting down the transmission in order to make time-consuming adjustments using expensive measuring instruments that only technicians and engineers know how to use. In addition, radio station directors are reluctant to lose audience by interrupting programming, even during the night.

It must also be considered that in many regions of the world, including central countries, there are not specialized personnel. Even in large cities, radio station engineers are disappearing [13]. Therefore, the task of maintaining good on-air audio quality or even knowing if a storm has damaged the antenna by measuring channel separation must be carried out by radio operators or remotely from the factory. Unfortunately, current FM modulation monitors do not provide information about the quality of the radiated sound, because they only measure RF parameters. Therefore, the audio distortion, dynamic range, or channel separation cannot be known without stopping the transmission and calling a specialized engineer. This is part of the problem this study had to solve.

The solution found is the replacement of the classic audio processor with a monitored audio processor. This replacement is not expensive because the price of this new equipment is the same as conventional audio processors. On the other hand, all radio stations change their processors due to obsolescence after 10 or 15 years of use. Therefore, in a decade or a little more, this new technology will definitively improve the sound quality of FM stereo worldwide. The proposed equipment, which has proven its effectiveness in hundreds of radio stations for the past 5 years, is based on a five-band DSP audio processor that includes a digital FM receiver, allowing the measurement of 24 RF transmission parameters and three sound quality parameters.

The interesting thing about this monitored processor idea is that it is controlled by a central microcomputer, allowing measurements to be performed at the receiver side by generating test signals in the transmitter, and it can be done synchronously and completely automatically at high speed without interrupting the transmission. Its action is almost imperceptible to the radio audience. A detail of the antenna of the five-band audio processor can be seen in Fig. 6.

This processor is controlled from any web browser on a PC or tablet connected to the radio's LAN network. From there, the measurement mode is activated, and a list of 24 RF parameters and three sound quality parameters is generated without interrupting the transmission. This list is displayed on the screen and can also be sent via email to the manufacturer for approval or advice on settings changes. But the most innovative feature is the audio quality measurement. This study will specifically look at stereo channel separation.

Fig. 7 shows a block diagram of the procedure used to measure channel separation while the radio is on the air without interrupting the transmission. It can be observed that only 50 ms are used to perform the operation of injecting a 1-kHz test signal into one channel while simultaneously cutting off the transmission in the other channel

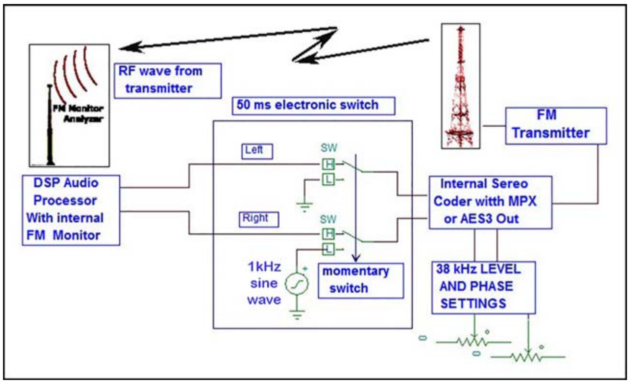

Fig. 7. Automatic measure of the stereo channel separation.

542 APC TECHNICAL REPORT - FM RECEIVER DATA - DATE(D/M/Y - 24hr):15/03/2018, 12:37:58

Frec sint:105.3 MHz
Valid: true
Stereo: true
MODULATION: 126.35 %
PILOT Mod: 6.5 %
RDS Mod : 4.8 %
RF Level Input : 92 dBµV
Multipath : 1 %
AUDIO L: 2.30 dB
AUDIO R: 2.30 dB
Carrier Offset :- 4.4 kHz
usn: -53 dB (ultra sonic noise)
assi200: -45 dB (SNR at 200 kHz offset relative to carrier)
lassi: -4 dB (Low Side Adjacent 100kHzoffset , SNR relative to carrier)
hassi: -9 dB (High Side Adjacent 100kHzoffset , SNR relative to carrier)
RDS BER (RDS- Bit Error Rate) : 0 %
RDS Data:
RT: Maroon 5 Wait (Videooficial)
PS:
PTY: 0
PI:FFEE

ON-AIR FM MEASUREMENT:
SEPARATION L>>R: 26.1 dB
SEPARATION R>>L: 21.42 dB
SEPARATION Label: Poor !!
THD+N : 0.7 % Fair

Fig. 8. RF and audio quality report of a radio station at Tierra del Fuego.

ON-AIR FM MEASUREMENT:
SEPARATION L>>R: 36.58 dB
SEPARATION R>>L: 36.72 dB
SEPARATION Label: Very Good
THD+N : 0.7 % Fair

Fig. 9. Audio quality report of the Fig. 8 radio station after a remote adjustment using a Monitored Audio Processor.

and measuring the 1-kHz residual through a bandpass filter. After a brief interval, the reverse operation is repeated.

The speed at which this is performed prevents the listener from detecting it, because during music playback, it is heard as just a few clicks resembling transmission noise. This allows the operator to shift the phase a couple of degrees and perform a new test. Once phase-optimized, they can adjust the gain until achieving a separation of more than 35 dB.

2.1. A. Real-Life Examples

Fig. 8 shows the report generated by a monitored processor of a radio that had just been installed in Tierra del Fuego, 3,000 km away from Buenos Aires, Argentina.

We have marked in red the reports on channel separation. It shows clear that despite having a processor with 65 dB of channel separation, a high-quality antenna with four stainless steel dipoles, and a good coaxial cable, the stereo separation is very poor as it falls far below the minimum of 30 dB required to authorize a radio station's on-air broadcast... 60 years ago. This situation has been repeated in 90% of radio station cases in most cities around the world. This occurs systematically according to these measurements conducted over a span of 5 years. In this case, from Buenos Aires, a remote adjustment was performed on a radio station in Tierra del Fuego, which took 3 minutes, and the results shown in Fig. 9 were obtained.

It can be seen that values above 35 dB of separation have been achieved, guaranteeing a perfect stereo sensation. The slightly elevated harmonic distortion is due to a transmitter that utilizes the old analog modulation technology.

Another example of a station with a high quality FM transmitter can be seen in Fig. 10. It can be noticed that an important impedance mismatch in the antenna was causing poor channel separation. After the adjustment, a separation of over 40 dB (Excellent) was achieved, which is at the receiver's discrimination limit and is identified as excellent.

This software version allows measuring the three basic audio quality parameters: Stereo Separation, total harmonic distortion (THD) distortion, and signal noise ratio (SNR).

It is interesting to clarify that all measurements are objective, expressed in decibels or percentages, and are important for engineers. But this work has taken into account that very often those who receive the report on the radio station do not have knowledge of electronic engineering and the values read do not mean anything to him. For that reason, a rating (Label) has been added that represents if the results are close or far from the quality level of the best FM equipment's today at the market.

This classification that can be seen in Table 3 allows the radio station to manage future investments to improve it. In the case of channel separation, the maximum rating has been taken as 40 dB, because no evidence has been found that increasing this value could be detected by the ear.

ON-AIR FM MEASUREMENT:	ON-AIR FM MEASUREMENT:
SEPARATION L>>R: 30.03 dB	SEPARATION L>>R: Excellent dB
SEPARATION R>>L: 16.88 dB	SEPARATION R>>L: Excellent dB
SEPARATION Label: Poor !!	**SEPARATION Label: Excellent!**
THD+N : 0.1 % Very Good	THD+N : 0.2 % Good
SNR L: 70.9 dBA Excellent!	SNR L: 70.6 dBA Excellent!
SNR R: 70.9 dBA Excellent	SNR R: 70.6 dBA Excellent!
Before adjustment	After adjustment

Fig. 10. Audio quality report of a radio station located at Pilar, Buenos Aires, measured with software version 2.xx.

Table 3 Label used for describing in words the audio quality level

LABEL NAME	THD % Distortion	Stereo separation, dB	Signal/Noise, dB
Excellent	<0.1%	>40 dB	>70 dB
Very Good	0.1–0.2	40–35	70–65
GOOD	0.2–0.5	35–30	65–60
FAIR	0.5–1	30–25	60–50
POOR	1–2	<25 dB	<50 dB
BAD	>2%	...	...

3. CONCLUSIONS

The analysis conducted, based on SPICE simulation, confirmed the reduction of stereo channel separation due to the coaxial cable and antenna impedance. It also demonstrates that it is the reactive part of the antenna load that is primarily responsible for the reduction in channel separation. Understanding the nature of this effect has led to the development of *monitored processor technology* that can measure the three essential parameters of sound quality without the need to interrupt program transmission.

All adjustments can be made at the radio station by non-specialized personnel or remotely. During the 5 years, the transmission quality of radio stations located in the United States, Europe, Latin America, Africa, Indonesia, Australia, and New Zealand have been improved using this technology.

The author would like to thank the people at Solidyne Labs for the implementation of this new technology; Alex Bonello for the initial design of the 542 processor and the AX10 computer core; and Nicolás Echave for his continued involvement in all phases of the design of software and hardware and his great contribution to the project. I am grateful to our partner Ricardo Sidoti for his excellent work in the final design and numerous ideas for manufacturing this equipment, and to Pablo Spinelli who has created the different sound presets and all the equipment videos and documentation.

5. REFERENCES

[1] K. Springer and J. Newsome, *Understanding DAB* (NAB, Science and Technology Department, Washington, DC, 1994).

[2] O. Bonello, *La Aventura del sonido y la música* (Editorial Dunken, Buenos Aires, Argentina, 2023), 2nd ed.

[3] J. Berg, C. Bustad, L. Jonsson, L. Mossberg, and D. Nyberg, "Perceived Audio Quality of Realistic FM and DAB+ Radio Broadcasting Systems," *J. Audio Eng. Soc.*, vol. 61, no. 10, pp. 755–777 (2013 Oct.).

[4] Solidyne, "Unidex Digital Mixers," https://www.solidynepro.com/unidex/ (accessed Mar. 28, 2023).[1]

[5] H.-P. Hahn, "Method for Noise Reduction in Receiving FM Stereo Signals," *J. Audio Eng. Soc.*, vol. 70, no. 3, pp. 208–238 (2022 Mar.). https://doi.org/10.17743/jaes.2021.0064

[6] Wikipedia, "Audicom: The First PC Radio Automation System," https://en.wikipedia.org/wiki/Audicom (accessed Apr. 10, 2022).

[7] J. T. Robinson and L. G. Whipple, "FCC Regulations Governing the Audio Fidelity Characteristics of FM Broadcast Equipment," *J. Audio Eng. Soc.*, vol. 16, no. 3, pp. 285–288 (1968 Jul.).

[8] NHK Japan, *FM Stereophonic Transmission Technique* (Nippon Hoso Kyokai, Tokyo, Japan, 1966).

[9] Telos Alliance, "Omnia.9 Specifications," https://www.telosalliance.com/radio-processing/radio-processors/omnia9 (accessed May 26, 2022).

[10] P. K. Onnigian, "FM Stereo Separation Degradation as a Function of Antenna System VSWR," presented at the *31st Convention of the Audio Engineering Society* (1966 Oct.), paper 502.

[11] O. J. Bonello, "Multiband Audio Processing and Its Influence on the Coverage Area of FM Stereo Transmission," *J. Audio Eng. Soc.*, vol. 55, no. 3, pp. 145–155 (2007 Mar.).

[12] G. W. Bartlett (Ed.), *NAB Engineering Handbook* (National Association of Broadcasters, Washington, DC, 1975), 6th ed.

[13] P. McLane, "*Radio Engineering in Crisis*," https://www.nautel.com/content/user_files/2020/05/Nautel-RW-ebook-Engineering-In-Crisis.pdf accessed Jan. 21, 2022.

[1] All websites cited in this reference list have been archived using the Internet Archive (https://archive.org/) on the associated date of access.

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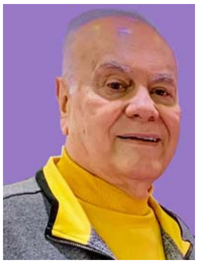

Oscar Bonello

Oscar Bonello was a Full Professor of Acoustics and Electroacoustic at the University of Buenos Aires. He has an MS Engineering degree from the same university. He founded *Solidyne Labs.* He is the author of the *Bonello Criteria* acoustic criterion, used worldwide for designing acoustics rooms. In 1988, he developed the world's first digital audio record/reproducing system working at PC that replaced the magnetic sound recording. He developed the first perceptual coding technology, ECAM. Mr. Bonello wrote about 150 technical articles and papers. He holds eleven Invention Patents. He gave numerous conferences at universities and published books. He is an AES Fellow, IEEE Senior Member, ASA Member, and founder Member of AdAA. He was granted with the First National Prize for technological innovation, the First Prize for innovation in radio broadcasting, the AdAA award for contributions in the field of acoustic, and the First Prize for high-tech audio products exportation.