# ADVANCES IN DIGITAL MEASUREMENT TECHNIQUES FOR FM BROADCAST

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## ABSTRACT

Advances in Digital Signal Processor (DSP), Receive Signal Processor (RSP) and accurate high-speed analog-digital converter (ADC) technology have made possible new concepts and techniques in measuring and analyzing signals used in FM Broadcast facilities. We will explore the advantages and limitations of this technology as it applies to FM Broadcast measurement and analysis. We will begin by exploring the differences between the architecture of traditional analog measurement equipment and the new digital model, and expand on the new theories and mathematics employed in making accurate measurements within the more open structure of digital equipment. The digital designs allow us to approach theoretical ideals in terms of filtering and accuracy, and this presentation will attempt to explain in solid terms the relevance and expectations of this capability in real-world applications.

One of the novel concepts to be explored is undersampling. We will explore the Shannon theorem, and present practical and mathematical evidence that the use of undersampling for signal analysis can produce excellent, accurate, results. The presentation will then provide a brief explanation of digital measurement techniques as they apply to each part of a broadcast FM signal – audio, composite/MPX, and final complex RF emissions.

## DEMODULATOR VS. RECEIVER

Let us first define the major difference between a receiver and a demodulator in terms of FM performance measurements and applications.

### Receiver

In general terms, a receiver has a high dynamic range, and accurate performance can be obtained at fairly low RF levels. Selectivity is high, but performance and accuracy are somewhat limited. A receiver accepts the entire FM band at its RF input.

### Demodulator

A demodulator has a low dynamic range (around 30dB), and a high RF level is needed to guarantee accurate performance (>-25dBm). It is a wide-band device with excellent accuracy. When used for signal analysis, only one RF carrier is allowed at the RF input.

## BENEFITS OF COMPLETELY DIGITAL EQUIPMENT

In order to understand why using a demodulator for FM signal analysis is innovative, and how it improves performance and accuracy, we have to first examine the current and most commonly used equipment architecture, and how digital techniques can improve not only the performance, but also add capabilities in terms of the number of instantaneous measurements.

### Analog receiver / measurement equipment

Up until today, FM demodulation and analysis has most commonly been done with analog components; the task is to demodulate a frequency somewhere between 87.9MHz to 108MHz, with channels spaced every 200kHz.

In the case of analog measurement products, the critical areas of the receiver are the RF mixer, RF filters and the demodulation component. We have to ensure certain things:

- That the tuner is able to tune every frequency in the FM band, without creating problems of tuning-frequency images
- That the tuner in the equipment is selective enough to extract only one channel (and not any portion of the adjacent channels)
- That the filter has a ±150kHz bandwidth around the selected frequency, with the flattest in-band frequency response possible

Long ago, the architecture for this type of receiver became standardized: analog mixers are used to blend the incoming frequencies against a tunable local oscillator, a technique known as heterodyning. The oscillator is tuned so that desired reception frequency is output on an Intermediate Frequency, or IF, of 10.7MHz. Ceramic filters are then used to separate the desired 300kHz-banwidth (or >400kHz-banwidth in the case of HD Radio tuners). But the 10.7MHz-filter is often a compromise between out-of-channel rejection, and flatness of the in-channel frequency response. It is very difficult to reduce in-channel ripple to less than 0.1dB when using analog components.

This inherent non-linearity of analog FM reception and demodulation inevitably adds a certain amount of noise and distortion to the output signal that is to be measured and analyzed. The tolerances of the analog components can also induce disparity in the performance from one measurement to the next, or the measurements made on different pieces of equipment. Regular calibration and adjustment of each stage of the equipment is also a typical need for these types of analog devices.

Today, there are many examples of measurement devices that digitize the signal for display on a graphic interface. In most cases, these devices digitize and convert the analog signal once it is completely demodulated (conversion of MPX signal or conversion of audio signals) and/or decoded. So, even if they are described as 'digital' equipment, they digitize the signal solely for the display of the levels. Measurements can be digitally displayed, but the underlying performance is limited by the analog demodulator, and analog stereo decoder.
As the analog to digital converter hardware can be very sensitive, most equipment has only one digitization chain: that single chain is used to convert only one signal at a time (or two, for audio). So to display deviation, pilot and RDS injection simultaneously, you would need 3 pieces of equipment stacked on your desk.

In the area between the older, completely analog devices and the newest, completely digital demodulators, we can find several other hybrid architectures, before we arrive at the best one in terms of performance and functions.

## Narrow band receiver

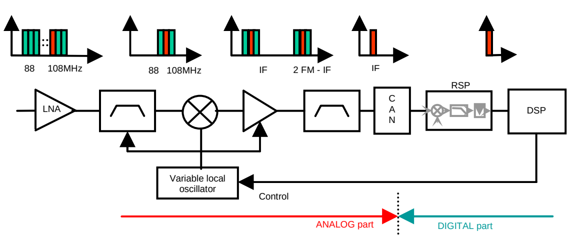


**Figure 1 : Narrow Band Receiver**

In this architecture, the first digital stage of the demodulation chain is a demodulation of the IF signal. In this example the analog tuner stage, from RF input to IF filter, is retained, along with the benefits and problems inherent to that design. Only one channel, translated from the desired frequency to the IF frequency (commonly 10.7MHz), is digitalized. RSPs and DSPs complete the FM-demodulation process and also stereo decoding. Measurements are extracted – except for the RF level which can not be measured because of Automatic Gain Control (AGC) used in the tuner stage. A digital filter can be incorporated into the RSP to increase out-of-band rejection.

This method allows simultaneous display of many different measurements. Digitalization in the IF stage eliminates analog FM demodulation non-linearity, noise and distortion, and demodulation does not depend on component tolerance.

The primary problem of this type of receiver is the analog RF stage which can add some noise and non-linearity. Some AM ripple will be created due to the IF-filter response, as its in-band frequency response can vary by several dB over the 400Khz bandwidth. Performance and accuracy will decrease, and will be less than optimal. As analog circuitry is still prevalent in the RF stage, it will require adjustment and regular calibration.

## Wide band receiver

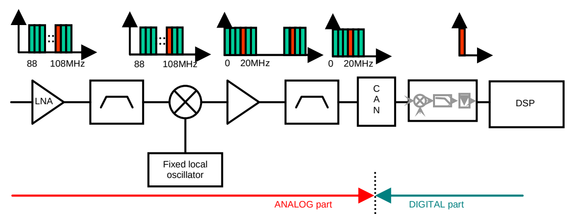


**Figure 2 : Wide Band Receiver**

With the new generation of accurate high-speed converters, we can now sample a signal over a large bandwidth with flat in-band response. This method is based on the sampling of the entire FM band, which is translated to 0-20.5MHz with a fixed frequency local oscillator (ex: 87.5MHz). An RSP, with its digital mixer, is used as the tuner and also as a digital filter to select the desired channel.

This method is more accurate than the previous one described. The use of an Intermediate Frequency and especially the IF filter are no longer needed, and the in-band filter response is flat (less than ±0.05 dB of ripple over ±150kHz), with high rejection (>90dB at ±200kHz).

In this method, almost every analog limit has disappeared: adjustments on RF stage are no longer needed, and FM demodulation noise, distortion and linearity are also perfect.

The weakness of this method is the need for a local oscillator, and the RF signal level which needs to be low enough to not saturate the analog to digital converter.

### Innovative architecture of demodulator

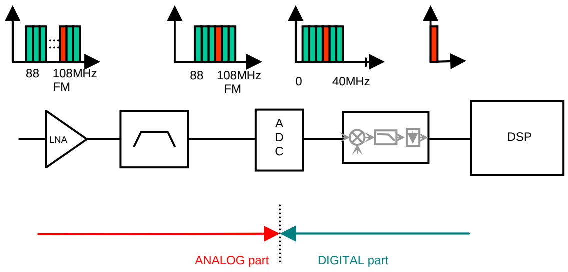


**Figure 3 : Innovative architecture of demodulator**

Some high speed AD converters are able to digitalize a bandwidth of up to 200MHz. The entire FM band can be converted without any mixing stage. Analog components are reduced to a single wide band RF filter (to reject signals outside the FM band), and overall hardware is simplified. The demodulator is reduced from 6 stages to 3, improving performance. The mixer (for tuning), channel filter, demodulation, and measurements: every thing is done in the digital realm.

To have the best accuracy, it is important to not saturate the A to D converter, and also to ensure that the level of your carrier is high enough. Calibration consists only of checking levels in the band and the exact frequency of the oscillator.

This architecture enables highly accurate RF, MPX, Pilot SCA, and Audio measurements to be displayed simultaneously and with perfect synchronization. Displays are not limited to a few bar graphs, or a single value on the front panel, but can now integrate multiple advanced RF measurements (Frequency, level, RF channel spectrum).

## 'IDEAL' VS. 'REAL' EQUIPMENT

Simulation of the perfect performance of a completely digital system is easy to do within the confines of advanced mathematic software. But some limitations unfortunately exist in the real world.

### Hardware limitation

In an analog receiver, the tuning accuracy is defined by the local oscillator used in the analog mixer. A slight drift of the oscillator will induce a measurable mistuning. Even if the signal is mistuned, the FM demodulation will still occur, but a DC signal will appear on the MPX signal, with a level proportional to the mistuning of the oscillator.

In a completely digital system, the heart is also the oscillator. To avoid synchronization troubles, you have to monitor the performance of all of your clocks. A drift of the oscillators could have some other dramatic consequence (glitches in demodulated signals …).

Thus the choice and performance of the oscillator has an important impact on some measurements (RF frequency …).

### Mathematics and DSP

Some mathematical operations are not native to DSP processors. Sine, cosine, tangent$^{-1}$, and square root calculations are approximations, with precision defined by the designer.

When implementing the algorithms, you have to have exact knowledge of what the required precision is; for example, you have to know this precision to ensure that your IIR[1] Filter will not diverge, because of rounding of calculations.

By using floating-point operations in a DSP, you avoid the scaling problems of a fixed-point DSP. Your filters are easy to import from your simulation, and the result is very close to that of your simulation.

## RSPS AND DSPS IN BROADCAST MEASUREMENT

### RSP – Receive Signal Processors

Commonly used in Wireless Infrastructure, this component includes a frequency translator (or digital mixer), and cascaded programmable digital filters.

Even if we could put frequency translator and digital filter into a DSP, an RSP is a device uniquely dedicated and better adapted to such computations, at a very high speed (80-100MHz). The use of a RSP in conjunction with a DSP, relieves the DSP of these basic mathematical manipulations.

The resulting performance, when these methods are used in digital measurement equipment is excellent: you can select any frequency you want in the FM band, within 0.1 Hz (theoretical value: it depends on your oscillator). The digital filters reject the out-of-band signal down to -90 dB, with less than 0.1 dB of in-band ripple…

### DSP performances

DSPs are ubiquitous in signal processing today. Wider availability of high-speed floating-point DSP chips allows fast implementation of complex algorithms, and incredible calculation power. A 1 Giga 32 bit Floating point operation (1 GFLOPS) DSP is common today, and allows nearly unlimited possibilities in broadcast measurement.

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[1] Infinite Impulse Response

## DSP possibilities

Imagine you want to analyze an entire high performance FM signal, while performing 4 FFT computations (RF, MPX, Audio Left, Audio Right), to check noise.

You also would like to simultaneously analyze or check every MPX signal parameter (Pilot, RDS, SCA, Audio), apply some specific filters (de-emphasis, weighted, un-weighted) and detectors (Peak-Peak/2, RMS, QuasiPeak), and matrix the signals while avoiding objectionable high-frequency levels in your headphone by applying a de-emphasis filter only for headphone output..

While you are checking this, your boss would also like to be sure that RF level and RF frequency are OK, and check that RDS data is being correctly set and sent (TMC, …), from the same equipment, without any interruption in your measurements.

That is what DSP can do - compute all of those measurements simultaneously, and make them all available. Then system just needs to be able do send the information over an IP network connection.

Some broadcast measurement equipment offers such simultaneous measurements, but you then need to have a powerful PC connected, and sometimes basic information such as RF level is not available.

## DSP Flexibility

You want to deploy new technologies (FMeXtra for example), and measure the performance? Your existing analog equipment does not have the correct filters, or it suffers interference from this new modulation? You need to buy new dedicated equipment, or retrofit new filters and hardware.

With DSP technology and a thorough knowledge of algorithms, just create the new filters, adjust the others if needed, and upload them into the equipment: That's all! Technology upgrades have never been so easy.

## DIGITAL MEASUREMENT TECHNIQUES

Now let us explore the mathematics that must be applied to this innovative architecture in order to generate useful data.

## Filtering

Just like in analog filtering, we can define two kinds of digital filters: filters without feedback, so-called Finite Impulse Response filters (FIRs) and the Infinite Impulse Response filters (IIRs) whose final output depends to a degree on feedback from previous output of the filter

An Nth-order FIR digital filter can be performed in the time domain using the following equation:

$$y[n] = \sum_{i=0}^{N} x[n-i]b[i]$$

**Equation 1: FIR filter computation**

- x : digital input signal derived by sampling the analog signal
- y : digital output
- b : the filter's impulse response.

By definition, there are N+1 terms to compute and add. The main advantage of using this kind of filtering is stability but if the value of N is too large (i.e. if the order of the filter is too high), computation time becomes an issue, since it is a convolution computation and demands a lot of processor speed and power.

An Nth-order IIR digital filter can be defined by a recursive time-domain equation as follows:

$$y[n] = \sum_{i=0}^{N} x[n-i]b[i] - \sum_{i=1}^{N} y[n-i]a[i]$$

**Equation 2: IIR filter computation**

Though the IIR filter can be regarded as an infinite order FIR, it has the advantage of having a very short recursive equation that is much easier to carry out. The main drawback of this approach is its potential instability. To ensure the stability of the equation and the accuracy of the resultant filter, one has to make sure that all of the poles are located within the unit circle. In digital filtering, we define the poles as simply the roots of the polynomial defined by vector a.

If any one of the poles approaches the unit circle, the resultant filtering could become inaccurate.

These types of filters are very convenient when it comes to common applications such as low-pass, high-pass or band-pass filters. Very high order filtering can be done without placing too much demand on the processors, and accuracy can be maintained by ensuring that the poles remain well within the unit circle.

The filter order is set depending on the desired steepness (sharpness) of the filtering.

**Example:**

For instance, let's filter a signal sampled at frequency of 50kHz. The analog signal is the combination of 2 sine waves at frequencies of 2.5kHz and 10kHz.

We want to remove the 10 kHz component from that signal. A low-pass FIR filter of order 62 is chosen.

Here is the b vector graph of this filter:

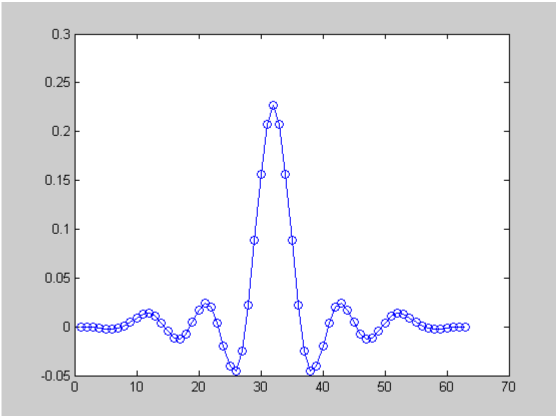


**Figure 4 : Time-realm filter impulse response**

Using a Discrete Fourier Transform of vector b one can easily compute its frequency response:

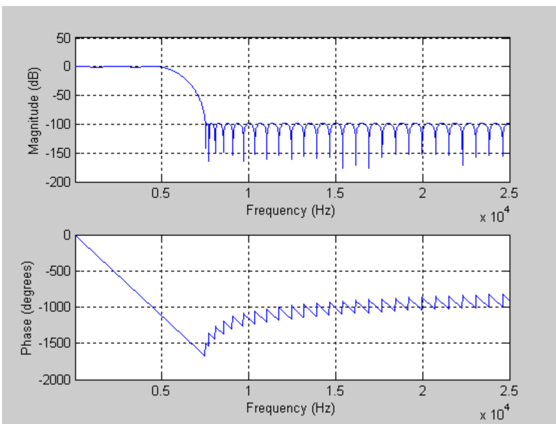



**Figure 5 : FIR Filter frequency and phase response**

It has a linear phase response in its bandwidth, which is an interesting characteristic of FIR filters.

The output (red) is derived by convoluting the filter function b[n] with the digitized input (blue). The result is a single sine wave at 2.5kHz, which lags the 2.5kHz component of the input signal slightly (0.62ms) due to the delay inherent to an FIR filter.

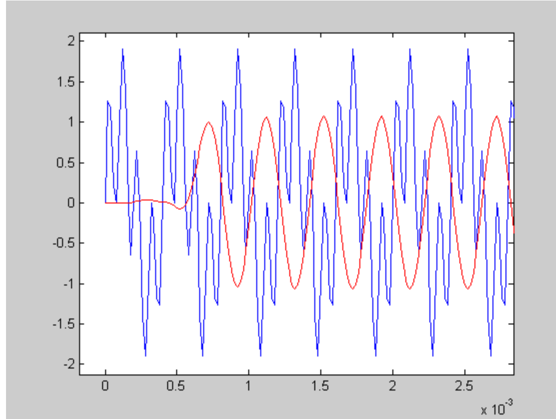


**Figure 6 : Result of digital filtering**

One can get a similar result using an IIR filter. Here are the filter's b and a vectors:

b = [ 1.0679e-004
-1.2500e-004
2.7007e-004
-1.3871e-004
2.6516e-004
-1.3871e-004
2.7007e-004
-1.2500e-004
1.0679e-004 ]

a=[ 1.0000e+000
-6.6277e+000
1.9942e+001
-3.5470e+001
4.0714e+001
-3.0848e+001
1.5060e+001
-4.3312e+000
5.6220e-001 ]

Now let's check the stability, by examining the positions of the poles in the unit circle.

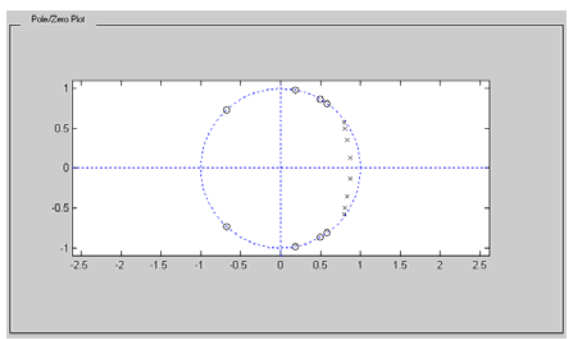


**Figure 7 : Poles (x) and zeroes (o)**

The filter in this example has 8 poles and zeroes which are inside the unity circle and not too close to it, so this filter is stable and the filtering will be accurate.

Z transforms, applied to the IIR's recursive equation, enable us to analyze its frequency response. A presentation of Z-transforms is beyond the scope of this paper, but the frequency response of the example filter can be graphed as below.

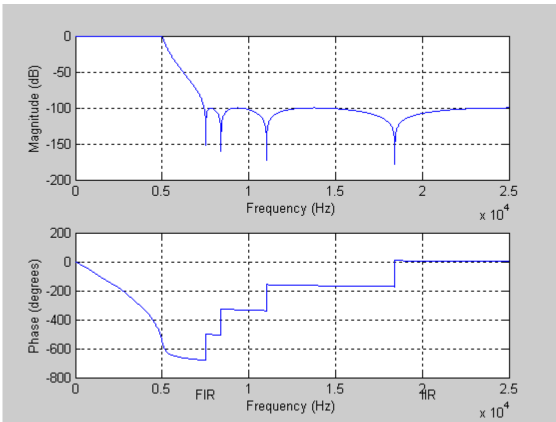



**Figure 8 : Result of IIR digital filtering**

This filter is as sharp as the previous FIR example, but is only an order of 8 as opposed to the FIR filter which had an order of 62.

Now let us compare the computation power required in both cases:

In this case an IIR solution is clearly more computationally efficient: only 17 multiply-add operations (MAC) are performed per sample to compute the output, whereas the FIR filter required over 60 such operations.

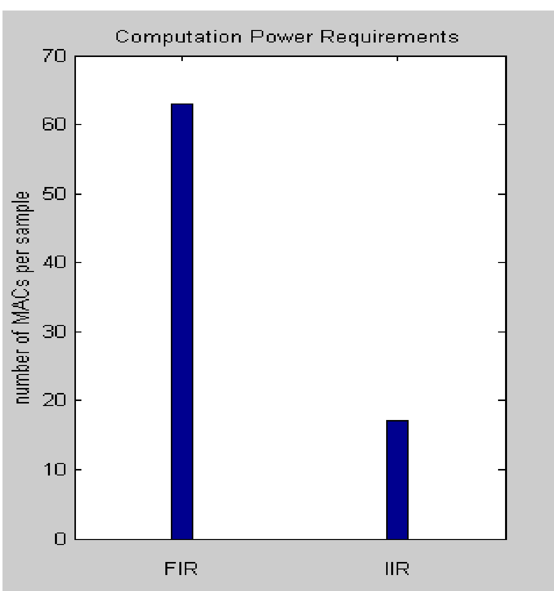



**Figure 9 : Computation power requirements**

## Advantages of Digital filtering:

Using FIR digital filtering, it is possible to approach theoretical ideals, provided that one uses an FIR order great enough. But this approach also requires more computation power. It is also possible to achieve excellent performance using IIR filtering while requiring far fewer calculations, but the accuracy of the filter must be tested and confirmed.

With digital filtering it is possible to design accurate reproducible filters. As a result, 2 channels can be processed, (for instance left and right audio), in an absolutely identical manner, unlike the case with analog filters. The response of the filters does not change with variations of temperature or over time. And of course no calibration is ever required.

## The FFT convolution approach to digital filtering:

As an alternative to traditional FIR filtering, it is possible to employ FFTs to carry out the filtering in the frequency domain. The filtering equation then becomes a simple product of FFTs in that domain.

Two FFT operations, an FFT and an inverse FFT, are necessary to enter and leave the frequency domain. This approach is more computationally efficient than FIR filtering if the order (N) is greater than approximately 60, depending on the hardware used. So this technique is strongly recommended in cases of high-order FIR filtering.

## Measuring the Instantaneous Derivative

Thanks to digital mixing, complex samples can be derived from a real-time analog signal which allows the calculation of the exact current instantaneous phase. In FM modulation, this phase equals the antiderivative of the instantaneous frequency.

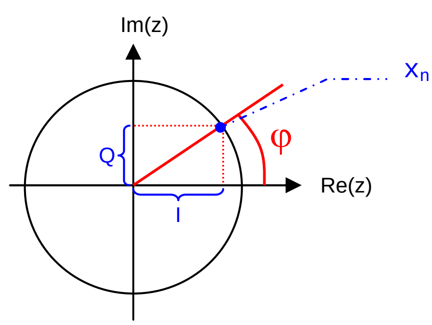
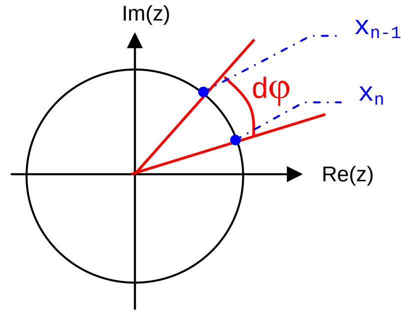



**Figure 10 : Complex sample representation**

By analyzing the variation of the phase between two samples, one can get an approximation of the instantaneous frequency value. If the sampling frequency is high enough, it is possible to derive the modulating signal.




$$U(n)=e^{j.\varphi n}.e^{-j.\varphi n-1} = e^{j.\underbrace{(\varphi n-\varphi n-1)}_{\text{Instantaneous phase derivative}}}$$

**Figure 11 : Complex sample representation**

Thus we can perform digital FM demodulation.

## Undersampling and Measurement Accuracy: the Shannon theorem

According to the Shannon theorem, in order to retrieve all the information of an analog signal, it must be sampled at a frequency at least twice that of its maximum frequency component.

For instance, let us consider an analog audio signal with a bandwidth of 15kHz. The sampling frequency should be greater than 30kHz, otherwise spectrum aliasing will occur. However, if one considers a signal of limited bandwidth, using a sampling frequency greater than 2*f_max is inefficient. By using a lower sampling frequency, it is possible to recover the entire FM band at a lower frequency, thanks to the fact that a digital signal has an Fs periodical spectrum. By using a sampling frequency of 80MHz, it is possible to recover the FM band and translate it to the 8 to 28MHz range.

When considering using undersampling, one must respect the following rules:

The occupied bandwidth of a narrow band signal must be less than half the sampling frequency and the entire bandwidth to be retrieved needs to fall between any multiple of the sampling frequency and that multiple plus one half the sampling frequency so that the signal is not affected by the sampling operation.

This is a generally accepted extension of the Shannon theorem.

Given that the spectrum of a digital signal is Fs-periodical, the relevant spectrum can be recovered between 0Hz and half the sampling frequency.

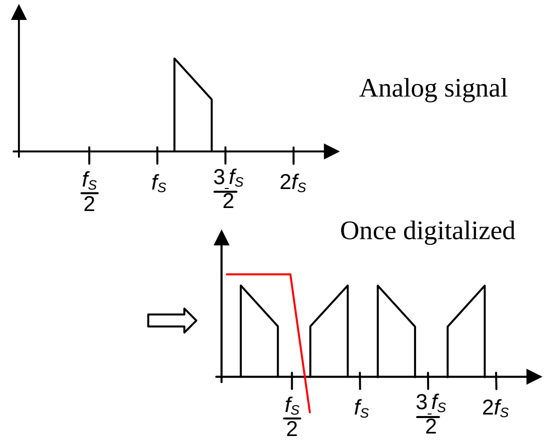



**Figure 12 : Complex sample representation**

Thus undersampling enables the processing of the signal at a lower frequency, without any additional demands or requiring any additional processing power.

In the case of the Navigator Modulation Analyzer, we chose an Analog to Digital Converter operating at 80MHz. Applying the technique of undersampling enables us to instantaneously recover the entire FM band and transpose it to 8 to 28MHz for analysis. In this particular case one can observe that this undersampling also acts as a frequency translator.

Using this method one can sample the entire FM band at once thanks to an 80MHz ADC provided that the incoming signal is limited to the FM band.

The technique of undersampling can introduce unwanted frequency components which may occur as a result of aliasing. It is a simple enough matter to eliminate those unwanted frequencies using an analog FM pass band filter on the input.

## SPECIFIC BROADCAST MEASUREMENTS AND EXPECTATIONS

### RF analysis

When checking the RF section of a transmitter, technicians must be able to test some critical parameters.

#### *RF Frequency stability*

When a transmitter emerges from the manufacturing process, technicians have to adjust the transmitter frequency. This adjustment requires a precise RF frequency meter to calibrate or validate an existing calibration. Required accuracy is within 100Hz.

After the transmitter is installed on site, we need to periodically check transmitter frequency. Even if a slight drift will not cause any perceptible problems with reception, it could be a sign of the aging of the transmitter. A large amount of drift is not acceptable, as it could disturb an adjacent channel, and must be avoided.

### RF power meter

To maintain the coverage area of the transmitter, the RF power level has to be checked. Aging of some stages of the transmitter can introduce AM modulation, or cause a slow continuous decrease of transmitted RF power.

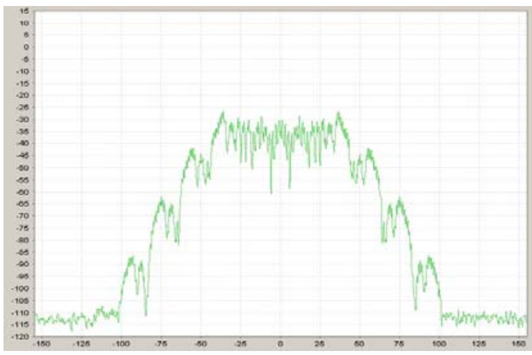


**Figure 13 : RF spectrum**

Measuring AM modulation on an FM signal requires a high speed measurement of the RF power, in order to detect the rapid fluctuations. Usually only specific equipment could be used – a traditional RF power meter which averages the signal will not be able to correctly measure the AM modulation.

### Spectrum analyzer to check out-of-band noise, and total deviation

When a new transmitter is tested, technicians have to check that undesired noise is not being generated by the transmitter. Noise could consist of harmonics of the carrier, or other anomalies in the system. Out-of-band emissions are completely forbidden as it could disturb an adjacent channel, but it is also necessary to check for emissions outside the FM band.

A spectrum analyzer can also be used to precisely adjust your total composite deviation. The carrier cancellation method is the most precise way to adjust your modulation. Generate a pure sine wave at a specific frequency (13587Hz if you want to configure 100% total deviation), bypass your stereo-encoder, and adjust the level of the sine wave as necessary to make the carrier disappear or be reduced as low as possible. You are now sure that your transmitter is correctly configured.

### Composite signal analysis

The composite signal is a multiplex of the combined audio channels, a pilot signal (used for stereo decoding), data stream (RDS), and other SCA.

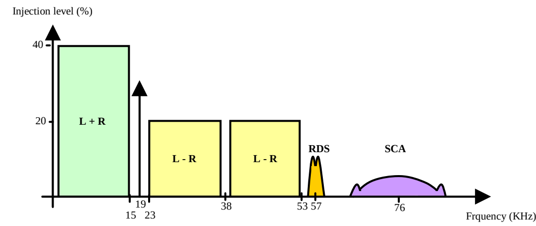



**Figure 14 : Multiplex description**

### Modulation

As total deviation is limited by regulation authorities, broadcasters have to respect the 75-kHz total deviation limit of their Multiplex signal (MPX).

In order to keep audio as loud as possible within these limits, the RDS and other modulation need to be set correctly: high enough to correctly decode RDS, and achieve lock on the Pilot, but not too high as to reduce the available modulation for the audio.

Adjustments of all of these signals are generally simplified by using a graphical display. Since changing one level changes the total deviation, simultaneously displaying every level with bar graphs, allows quick and accurate sound processor settings.

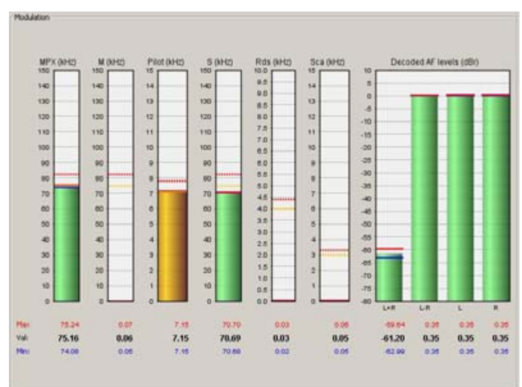


**Figure 15 : Bar graph display**

### Spectrum

To analyze certain problems, such as an unstable pilot or noisy audio output, spectrum analysis can be the best tool to gather information. What could be easier to see than a moving noise floor, or a Pilot signal buried in audio?

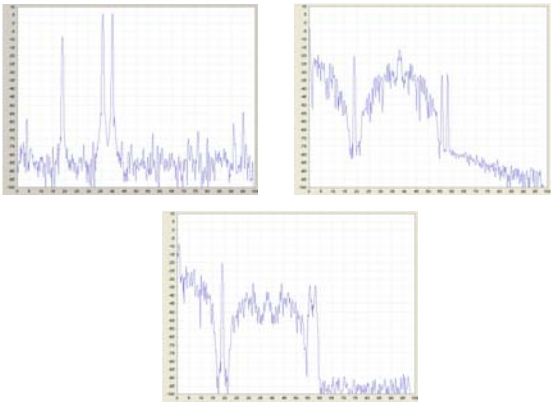


Figure 16 : MPX spectrums

Unstable levels or a bad noise floor can indicate a sound processor problem (second figure), such as bad audio filters or incorrect sound processor settings. High-frequencies in the audio signal can disturb the Pilot or RDS. You can easily check for this on the MPX spectrum display. Pilot and RDS have to be correctly isolated (correct in last figure).

## RDS decoder & analyzer

Once the RDS injection level is correctly set, it is useful to check that the RDS data is being sent correctly.

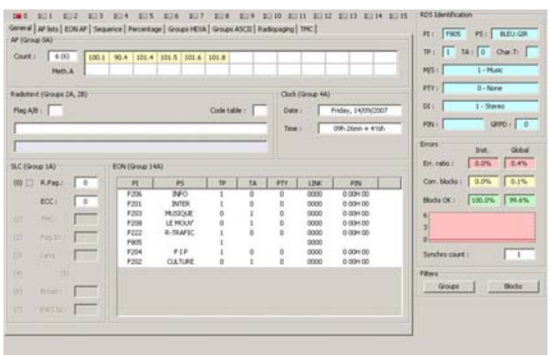


Figure 17 : RDS decoder

You need to use a RDS decoder and RDS analyzer to display data being received. An efficient RDS analyzer should display the basic RDS information (PI code, PS, AF) and specific application data (TMC, paging).

## Audio signal analysis

The main audio measurements of the transmission chain consist of checking the frequency response, distortion, and also separation. With new digital studio and transmitter equipment, these parameters have never been so good, and the performance of the old analog test equipment is sometimes worse than the broadcast chain being checked; the performance of the measurement equipment needs to be perfect. What could be better than measuring digital equipment with digital equipment?

Digital techniques can help us to have the best separation, distortion and accuracy in station performance and measurements.

## Filter response

The (left + right) and (left – right) audio signals are extracted from the composite signal using FFT-convolution pass-band/low-pass filters as seen before.

This allows for steep, accurate and phase-limited filtering, with a perfect control of the cut-off frequency. As a result the audio channels are not altered by this retrieval process, and the ensuing audio measurements will only measure the transmitter's defects.

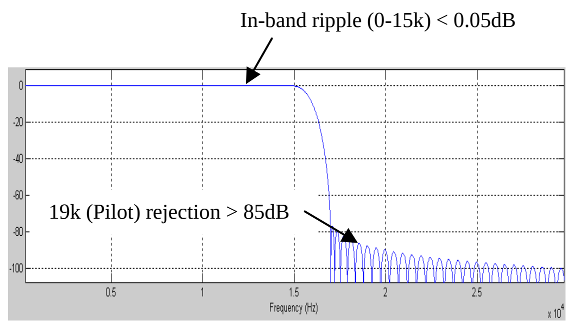



Figure 18 : Audio digital filter

## Distortion

The aim of the distortion (THD) measurement is to check for non linearities in the chain: saturation, clipping, compression…

There are two kinds of THD measurements: THD and THD+N on both audio channels.

Basically a THD measurement consists in comparing the level of the fundamental signal to the level of the remaining harmonic frequency components.

$$THD = \sqrt{\frac{H_2^2 + H_3^2 + \ldots + H_N^2}{H_1^2}}$$

THD+N compares the fundamental to the entire spectrum minus the fundamental, thus it also takes into account noise.

$$THD + n = \sqrt{\frac{H_2^2 + H_3^2 + \ldots + H_N^2 + n^2}{H_1^2}}$$

A THD compares the fundamental to all the other harmonics of superior order.

An easy way to assess distortion is to use an FFT computation, however it lacks accuracy, especially in the case of low-frequency fundamentals if no under sampling is carried out.

An accurate distortion measurement can be performed in two steps:

First, one computes the frequency of the fundamental signal, using for instance an array of Discrete Fourier Transforms (DFT) around the fundamentals' assessed frequency. Using a dichotomic algorithm one can fine-tune this frequency measurement.

Second, one computes the harmonics' levels using DFTs at frequencies that are multiples of the fundamentals'.

A THD+N measurement also requires measurement of the power in the audio channel; notch-filtered to remove the fundamental. A notch filter can be a mere 2nd-order IIR digital filter.

Thanks to this all-digital approach, we only measure the distortion caused by the transmitter. And it is measured very accurately.

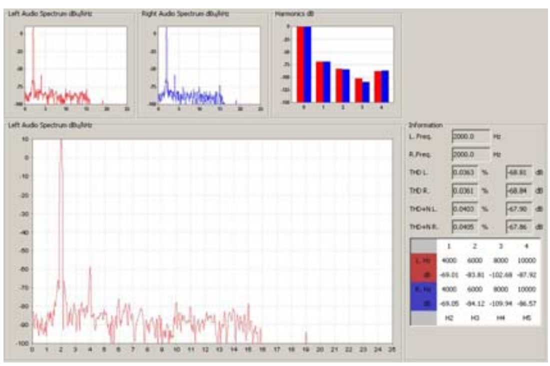


**Figure 19 : Distortion display**

### *Audio separation*

The last critical point is separation. When transmitting right and left channels, one expects than right channel will arrive in right loudspeaker without being disturbed by left channel.

In transmission chain, cross-talk can be created by bad synchronization of Pilot and 38kHz (center frequency of L-R channel), or poor filters. If this phase relationship is bad, the receiver can not correctly frequency-translate the (Left – Right) signal, and then recombination of Multiplex signals will not correctly produce the discrete Left and Right audio channels.

The performance of the demodulator depends on two major parameters: flatness of the audio filters and accuracy of Pilot recovery. As described before, the flatness of digital filters is more than sufficient to ensure no noise is added. Digital Pilot recovery is based on a Digital Phase Lock Loop (DPLL). The 38kHz center of the (Left – Right) signal is easily computed with an arithmetic formula, and the frequency-translation of (Left - Right) becomes easy.

Once again an all-digital technique proves to be the best approach: it does not add any significant cross-talk, so what is measured is only the performance of transmitter; and it is measured precisely.

## CONCLUSION

In conclusion, we can safely say that Moore's law, about the capabilities of integrated circuits improving at an exponential rate, has as much impact on the equipment that we use for broadcast measurements and analysis as it does for cellular phones and laptop computers.

The capabilities of these new circuits, along with innovative and advanced application of advanced mathematics, give us the ability to observe the signal components of an FM broadcast chain in new and better ways, and with higher and higher levels of accuracy. Where this will lead us in the future we can only imagine, but it will certainly advance the quality and reliability of the connection between the broadcaster and the public, and that will undeniably be good for all of us.