Theremino System

Minimize the FWHM

Methods for the reduction of the rows width with NaI (Tl), Csl (Tl) and BGO scintillator crystals

Thanks

First of all you need to explain why in this document use the plural form "we" and it is important to highlight that the results obtained are the result of joint work. Many have contributed to the research, first of all **Alessio Giusti** which has been a constant source of reference and which has played a large part of practical experiments. But also **Roberto Cena** who helped to set up the basics of the system Theremino, without forgetting **Andrea Bosi** who helped with software testing, interesting ideas and useful suggestions. But also **Steven Sesselman** of **Gamma Spectacular** which has stimulated the improvement of the FWHM. The last year has also added **Nobu Komagata** that is transforming ThereminoMCA into an ultra-professional application. Then there are hundreds of **experimenters** that for privacy reasons, we can not list here, but they have contributed a large number of tests, and have sent many pictures and suggestions. Finally, there is **Livio Cicala** (I), which has done most of the "dirty work" by synthesizing the ideas and advices of all.

Abstract

This document explains the methods that we have identified in the years 2012 and 2013 to minimize the value of FWHM of the normal crystal scintillators available with reasonable prices. These crystals are sometimes preferable to systems HPGe (hyper-pure germanium) is not only the price but also to be able to use light and portable systems (without tank of liquid nitrogen)

The scintillator crystals normally used, Nai (Tl), CsI (Tl) and BGO, have a poor resolution, almost to the limit of the useful. With these systems it is difficult to distinguish the lines of isotopes with similar energies. In addition, the broadening of lines creates a noise carpet that makes disappear completely the lines due to isotopes with low concentration and activity.

Even small increases in resolution produce large benefits on the visibility of the lines then we'll start the show with the remarkable results achieved in order to proceed with the explanation of the techniques that allowed them.

Resolution of crystals and improvements

The documentation available on the net until now considered as minimum the following FWHM values ***(1)***:

- `NaI (Tl) = 7%`
- `CsI (Tl) = 8%`
- `BGO      = 11%`

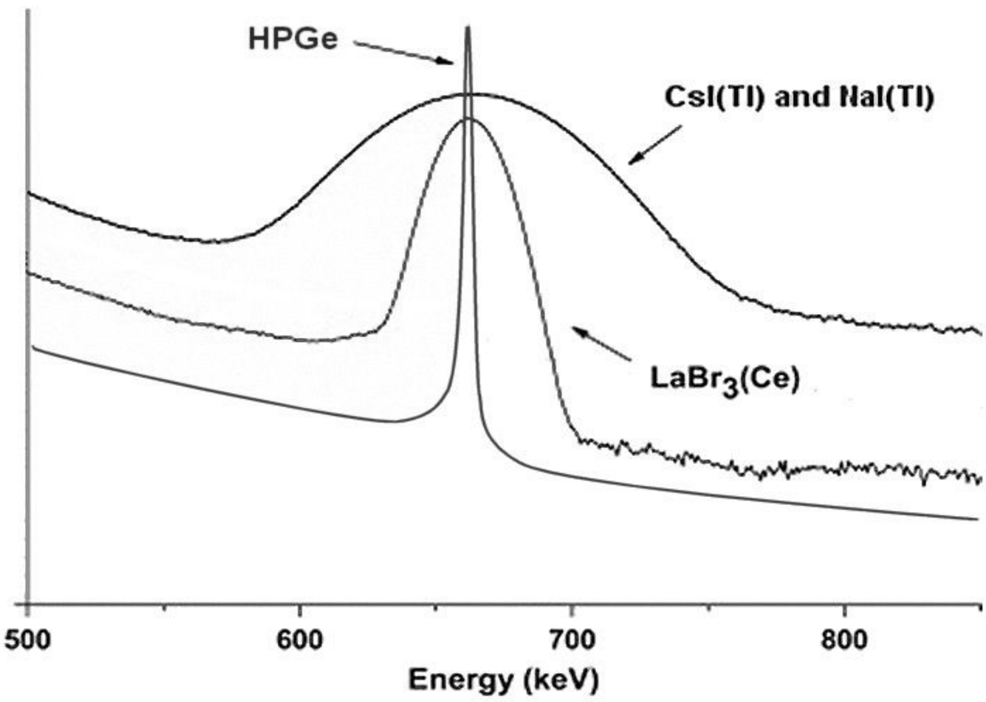
If you apply all the techniques that we present in this paper can reach the following values (and even something better according to the data of some investigators):

- `NaI (Tl) = 5%`
- `CsI (Tl) = 6%`
- `BGO      = 8%`

(Note 1) *On this page and throughout this document, the FWHM values are always referred to as Cs-137 and 662 keV are approximate values, do not take them as natural constants. Depending on the software, hardware, crystal and how it is coupled with the PMT, the actual values may differ from those shown, even by one percentage point.*

The resolutions compared

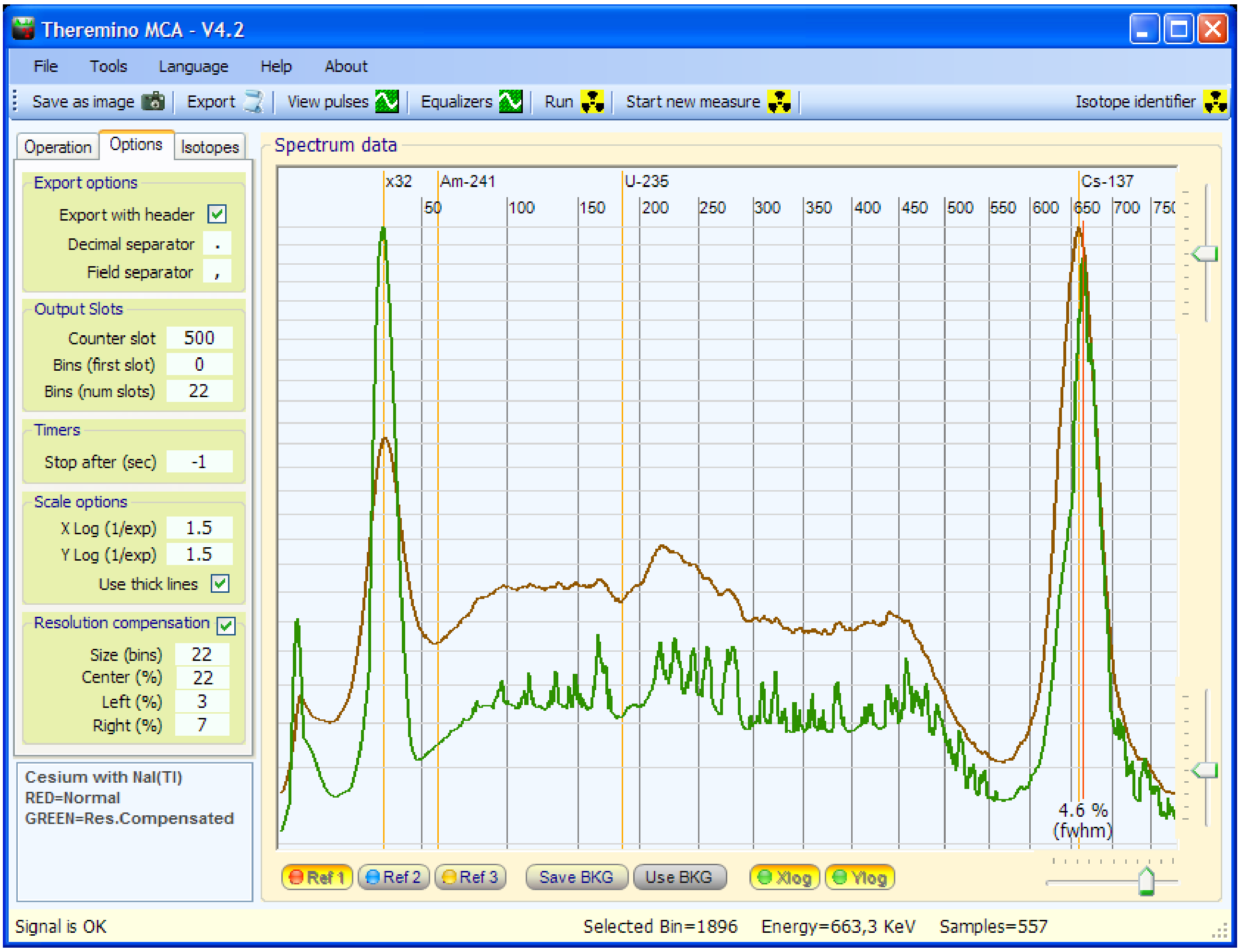
This is the approximate appearance of the lines produced by the main types of detectors.



The techniques we propose allow to **get with normal scintillators NaI (Tl), approximately the same resolution as the much more expensive LaBr3 (Ce)** Theoretically our software with a probe LaBr3 could get to the 2% FWHM and, with future improvements to the compensator for a resolution that we are planning, we could go to a incredible 1.5%

Decrease the value of FWHM in addition to tighten the lines makes sprout from the ground felt even lower rows, as shown in the images of the next three pages.

An example with the Cesium

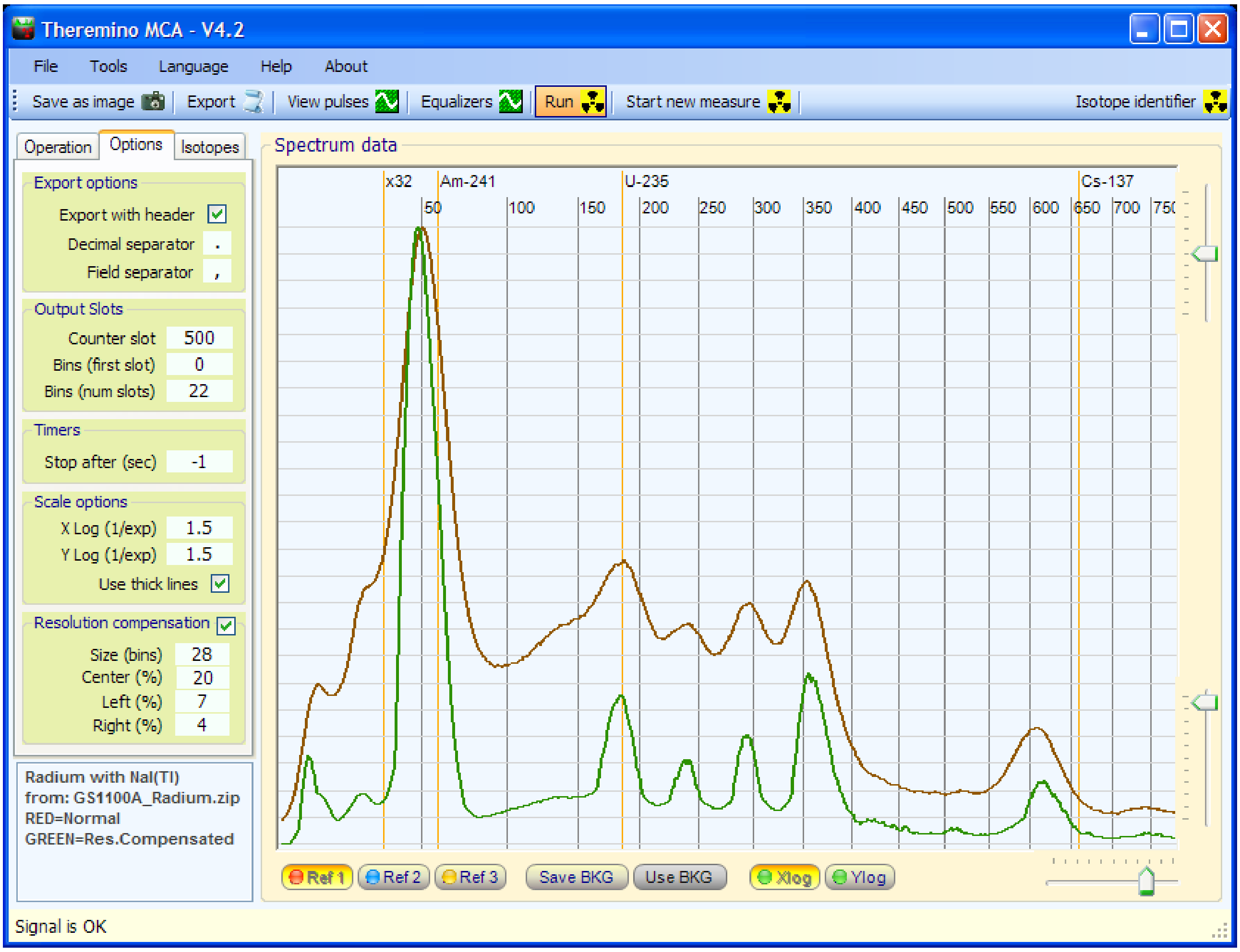


Here you see a sample of Cs-137 measured with the best probe Nai (Tl) at our disposal (red line) The value of FWHM relative (not shown in this graph) was 6%

Then without moving the probe and the sample, but only by enabling the algorithm "Resolution compensation" has got the chart with a FWHM of 4.6% (Green line)

The control panel of the "Resolution compensation" is visible in the left column, below.

An example with the Radio



Here you see a sample of Ra-226 measured with a probe from the mediocre resolution (red line)

The same data file was then read again but with the algorithm "Compensating resolution" activated (green line)

Here we have not enabled the meter FWHM, the FWHM values of the Radium are always very high and do not normally use.

How to "pick up" two points of FWHM

For details on the techniques from 1 to 16 see Appendices 1 to 16

1) Wiring of the PMT at low current, single capacitor wiring and nothing strange with zener and transistor.

2) Power from USB, no separate power supplies, no loop masses.

3) Filter signal conditioning that creates pulses of about 100 uS with a rounded tip (Gaussian)

4) Deleting the zero-pole (not to swing the zero line)

5) Pre signal amplification of about 30dB after the filter and before dell'all'ADC

6) Maniacal care lowering the noise (max 500uV pp, then nothing circuits "crowded" and full of wires)

7) High-pass filter before the ADC (eliminates low frequency noise and creates the bipolar pulses)

8) 16-bit ADC (for our approach "all software" 14 bits are the bare minimum)

9) High sampling frequency (at least 20 samples for each pulse)

10) From here on, all high-level software. Nothing DSP, embedded systems, and hardwarume varied.

11) Precise measurement of the peak width (best of SampleHold hardware on pulses "difficult")

12) offset compensation of the zero line with BaseLine Restoring.

13) Identification of PileUp.

14) Identification of pulses too "noisy" (noise, ringing, or pileup zero line inaccurate)

15) Finding the number, amount and type of errors and display control of the shape of the pulses.

16) Compensation of the resolution of the crystals.

The techniques shown by (1) to (15) have allowed to reduce the FWHM of about a point*. Until the end of 2012 (prior to version 4.0 ThereminoMCA) they were able to get to the 6% FWHM and* ***no further improvement seemed possible*** *(Not very different from that stated in the literature)*

The technique (16) "Compensation of the resolution of the crystals," *from version 4.0 onwards (beginning of 2013) has produced alone* ***another improvement of a full point.***

Which of these techniques affect the FWHM

All points listed affect the value of FWHM.

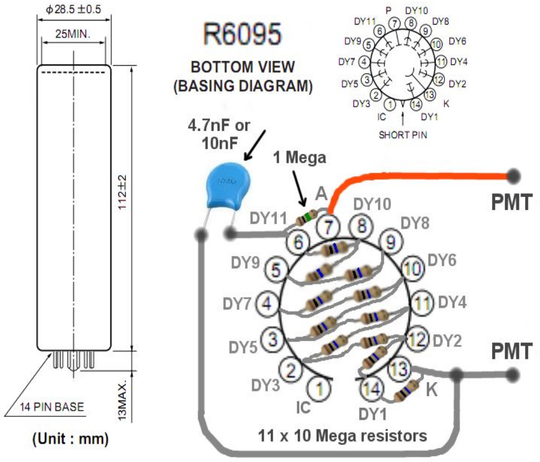
The documentation "theoretical" might suggest that only some of these techniques to be related to the value of FWHM but in practice (with real impulses and disturbed) the minimum FWHM is obtained by optimizing the entire chain, from the gamma ray up to the chart.

Although the details are not directly connected to the FWHM influence, as preparatory for the subsequent techniques, which otherwise could not function well.

To prove these claims just disable one by one, the various areas of software ThereminoMCA and see the changes in FWHM that follow. For noise, the number of bits of the ADC and other factors that are not modifiable by software methods exist external to ThereminoMCA. For example add increasing amounts of white noise with the application Theremino_AudioExamples.

Appendix 1 - Wiring of the PMT

If you do not follow these tips the noise increases , the sensitive measurement of the pulse width is disturbed , and the value of FWHM increases.



Low current

Only with low current PMT (total resistance above 100 Mega ohm) you can filter the power supply in an appropriate manner and reduce noise to a minimum.

But some manufacturers, such as Scionix, they use very low value resistors (470 ohm) to minimize the measurement non-linearity of up to 50,000 CPS and beyond.

If you were to measure 50000 CPS our advice would not be to linearize the sensor but to get away as soon as possible.

Measuring radioactive sources with high impedance probes

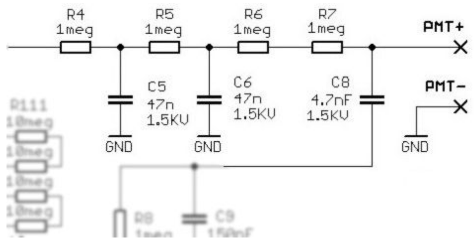
Even our probes at low current can measure very "hot" sources with a perfectly linear spectrum, proceed as follows:

1) Place the probe close to the sample and measure the CPS quickly (a few seconds) in this measurement gives the value of a count and linearity does not count.

2) It goes away just enough to take the CPS less than 500 (even if the source is several meters large) and this is good for both the probe and for the health of those measures .

3) Wait for the complement of the spectrum according to the safety distance. The resulting spectrum will be much more linear than that which would be obtained with 50000 CPS and probe with 470k resistors

High Voltage filtering capacitors



The double filter cell of PmtAdapter already uses abundant and rare capacitors 47 nF to 1500 volts. For a PMT with 470k resistors these capacitors should be a microfarad .

It's not just cost , capacitors from 1uF to 1500Volt would be huge and also very dangerous .

No "strange" components

When connecting the PMT there must be components such as transistors and zener . They are designed to make a linear response also in the area close to saturation, but with the weak signals of interest to us we are far from that condition and are useless . On the other hand create noise, higher power consumption and defects of all kinds.

Errors of this kind (eg linearize at all costs) are typical of the design " school " that sometimes overstates the importance of some details at the expense of overall performance.

Appendix 2 - Power supply from USB

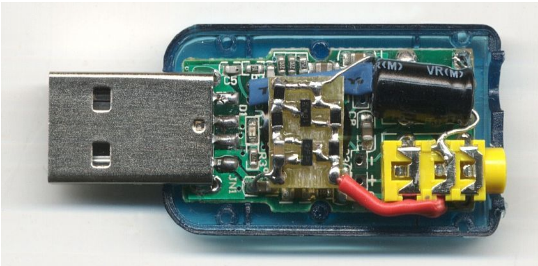
If you do not follow these tips the noise increases, it disturbs the amplitudes of the pulses, and increases the value of FWHM

No separate 5 Volt power supply

If you do not use the power from the USB, which follows the signal on the same cable, you inevitably create ground loops that cause an increase in noise unacceptable.

Using a sound card modified with power supply and high pass filter

Only by changing a sound card as indicated in the file PmtAdapters you get to bring the power on the same cable signal, thus eliminating ground loops. Of course you can not make these changes on the sound card on the motherboard and not even an expensive SoundBlaster.



There is fooled by their low cost, these coupons sample at 192 KHz and have a signal to noise ratio of over 100 dB.

For what serves us are perfect and even better than many cards from 100 Euro and more.

No "professional" power supplies

Connected power supply from the mains, causing inevitable noise at 50Hz on the delicate connection that carries the signal to the sound card.

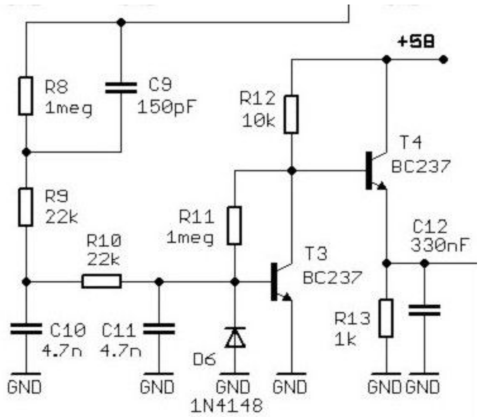
In addition, these power supplies are made for completely different uses, providing 1mA against the few uA that we need, but at least they have a noise intolerably high. *According to the manufacturer's specifications this power supply has* ***a ripple of "0:01% of full scale"*** *then: 10KV / 100 * 0.01 =* ***1 Volt pep (ten thousand times greater than our***)

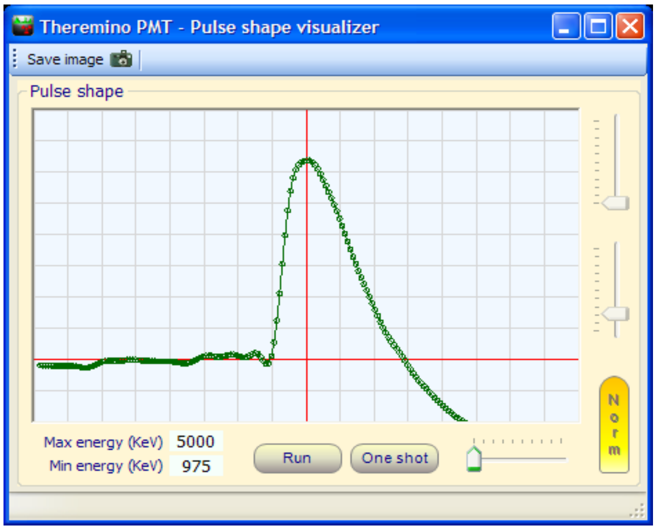
This is a beautiful PS355 from $ 2475, Stanford Research Systems from 1 mA to 10 KV (which are 10 watts)

Great for capital punishment, a little 'less for a tube PMT, just mistakes "kaputt" (PMT kaput, or human kaputt, depending on the error that you do)

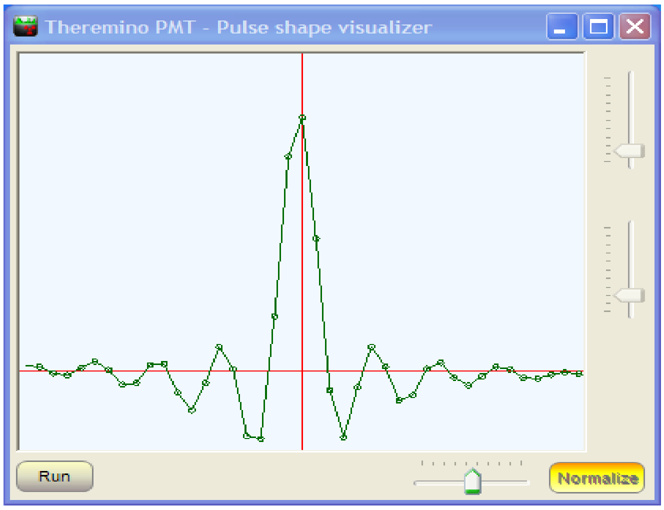
Appendix 3 - The signal conditioning

The filter signal conditioning creates pulses of about 100 uS with a rounded tip (Gaussian)





The pulse stretching to more than 100 uS allows you to sample the pulse with many points and make precise measurements.

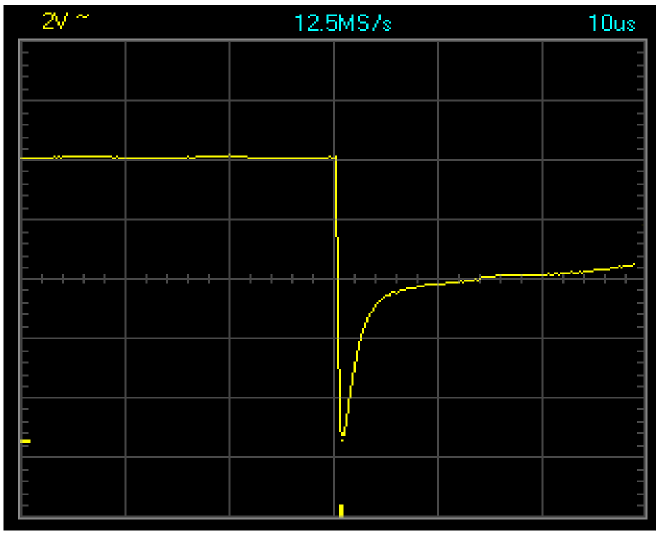


In the picture to the right you see instead a very narrow pulse, 5 uS (the horizontal scale is enlarged twenty times)

Note that you can not find the height of its tip with precision.

A pulse with steep edges also generates a lot of "ringing" (the undulations before and after the pulse) and the "ringing" makes it difficult to accurately measure the position of the zero.

To measure this pulse with the same precision as the previous image you should use an ADC with frequency twenty times greater than ours, then about 4 MHz



But here you can see the original impulse of a PMT without filters enlargement.

Many MCA hardware uses these impulses directly thinking that it is enough to use a fast ADC to measure narrow pulses, this is true but it should be faster?

The pulse is narrow and the tip is spiked (not with appearance Gaussian) 1 uS appears here, but the part of the needle tip should be 0.5 uS or not.

To have the same our accuracy it would take a 40 MHz ADC (our 198 KHz to 100 divided by 0.5 uS uS) But no system for gamma spectrometry using ADC 40MHz!

Appendix 4 - Cancellation of the pole "zero"

The pole zero cancellation minimizes the rocking of the zero line and facilitated the work of the Baseline Restorer.

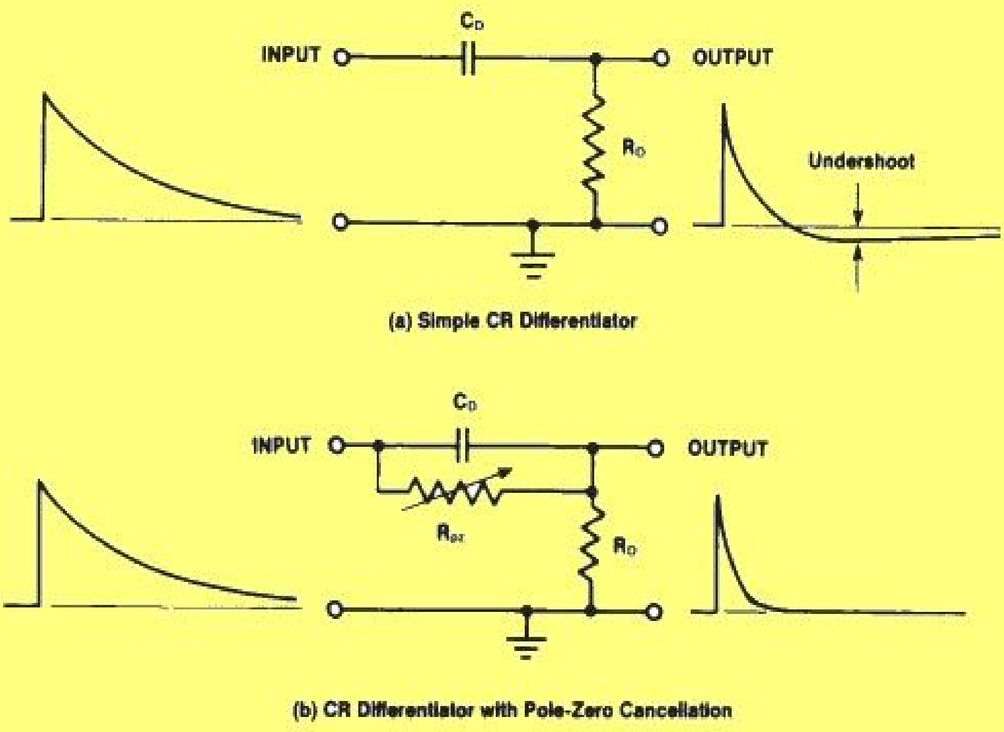


Fig. 12. The Benefit of Pole-Zero Cancellation.

The "Pole-Zero Cancellation", well described in the literature, eliminates the "Undershoot" and produces a pulse that goes to zero in the shortest possible time.

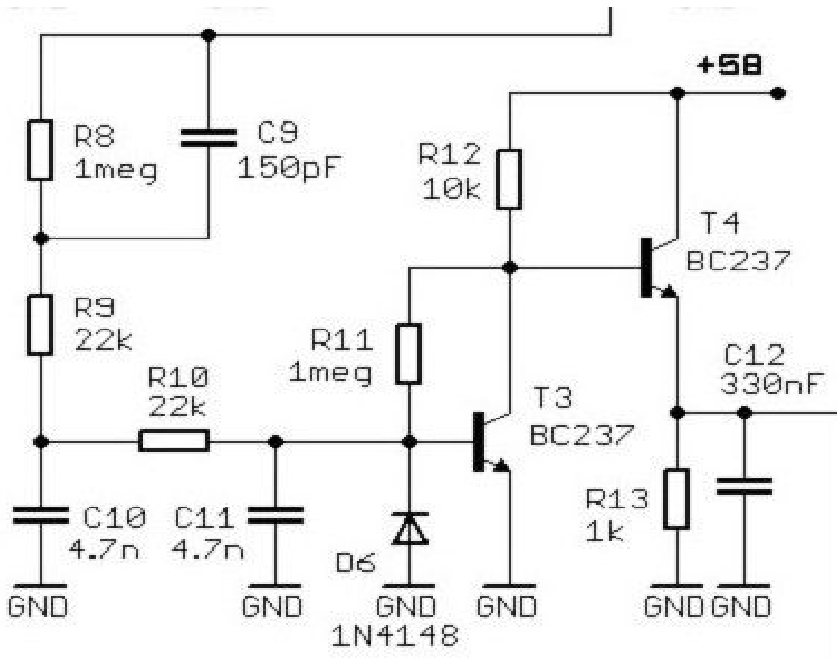
Having no pulse undershoot is a prerequisite to the high-pass filter can then produce pulses perfectly "bipolar"

The high-pass filter and bipolar pulses will be described in the following pages.

Appendix 5 - Preamp

The signal from the PMT after passing the filter and before going to the ADC must be amplified by about 30 dB and lowered impedance.

Without the preamp you might not get the full dynamics of the ADC, the noise would increase and would increase the value of FWHM



The transistor T3 amplifies the signal that has gone down a lot in the filter.

The transistor T4 and the capacitor 330nF lower the output impedance, minimizing the noise that the signal conductor can collect for coupling with the power lead and external disturbances caused by the mains.

This preamplifier slides the signal to the most suitable level for the ADC.

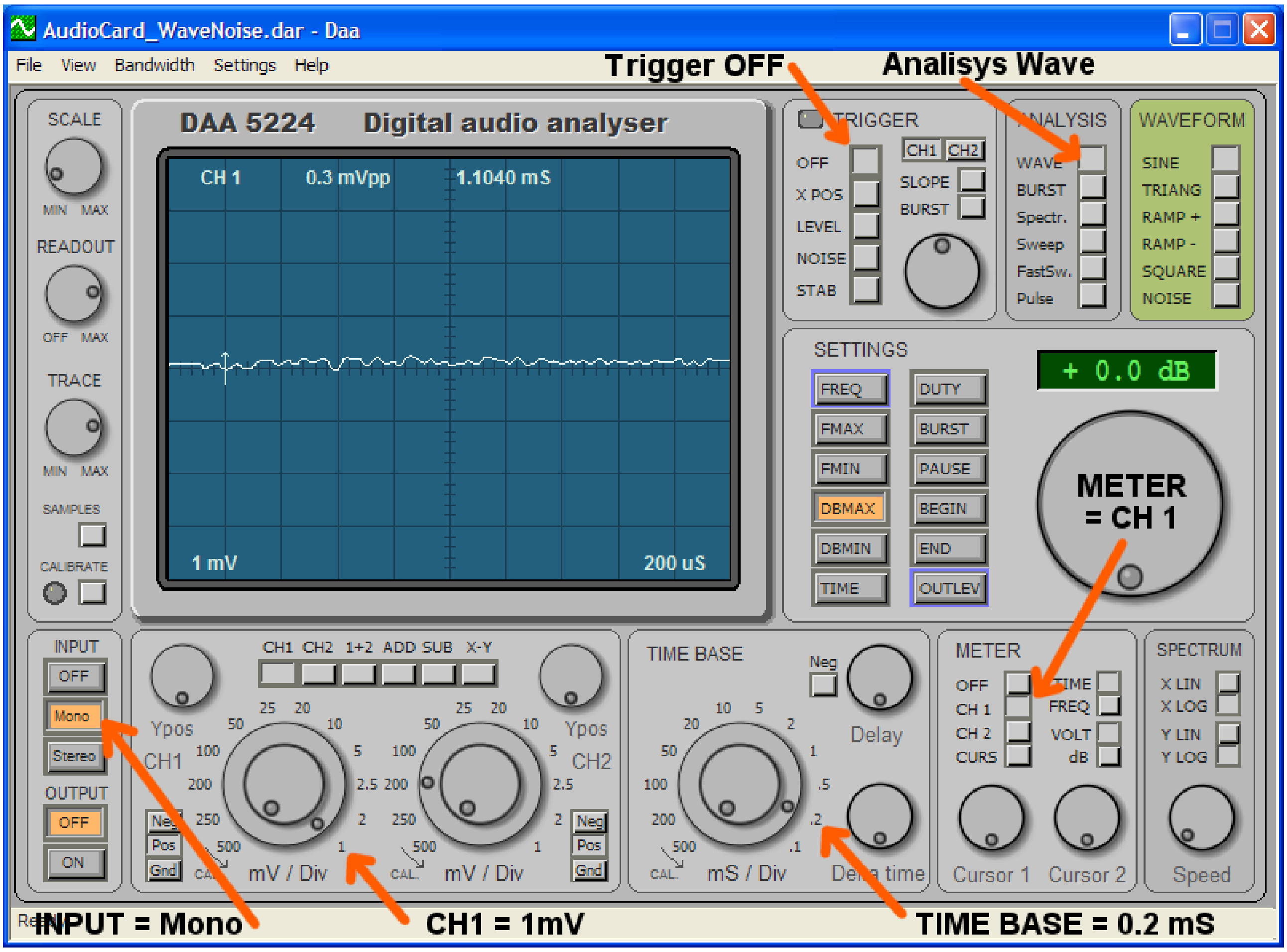
If the signal exceeds 1 Volt ADC saturated if it is too low you lose dynamics and increases the noise.

With proper adjustment of the voltage value, and then the amplification of the PMT, it maximizes the dynamic and you get the maximum ratio between signal and noise as possible (more than 100 dB)

Appendix 6 - The noise

As explained repeatedly requires great care in lowering the noise, so no "circuits busy" and full of wires and no DSP, escalators or other gadgets hardware.

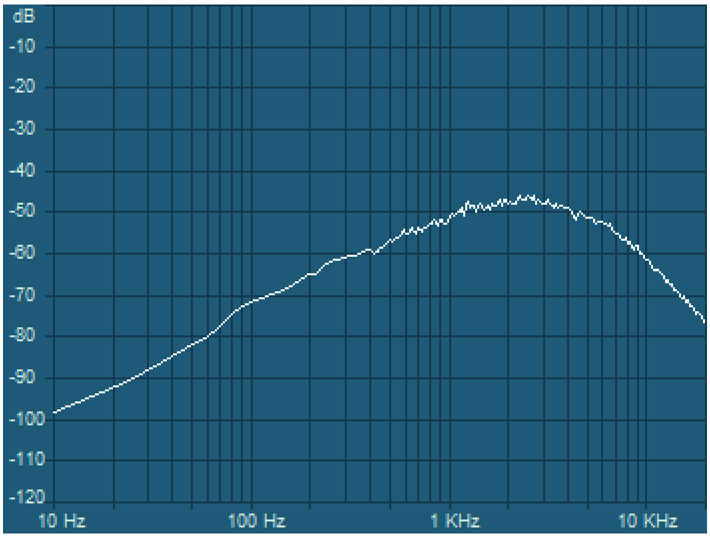
The higher the noise, the greater the uncertainty in the measurement of the pulse width and the greater the FWHM.



If you follow our directions the noise should not exceed 500uV PP, better if you get to 300 uV pp as in this image.

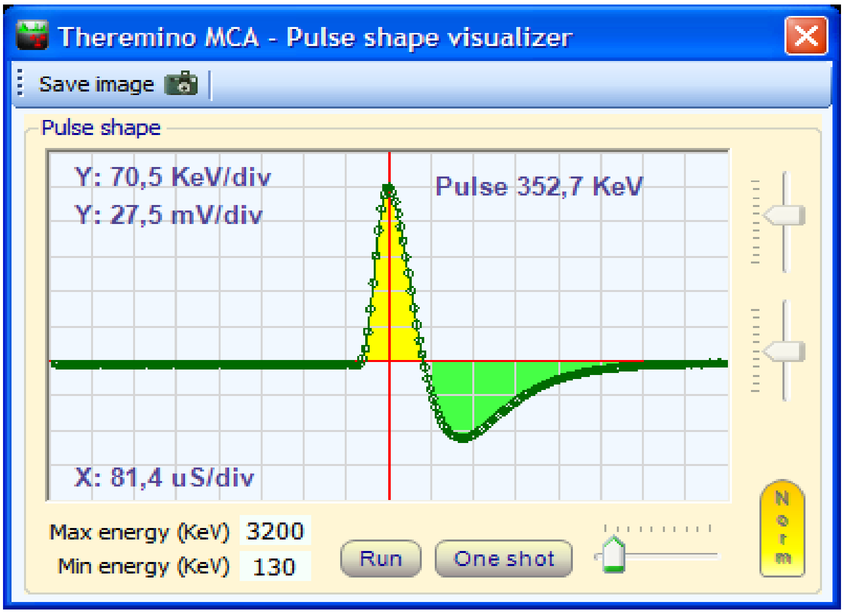
The voltage measurements are referred to a calibration with which ADC saturated at + / - 1 volt (approximately the conditions that you will get if you adjust the mixer and disables the ADC as shown in the file PMtAdapters.pdf)

Appendix 7 - The High Pass Filter



This is the average spectrum of the pulses after the conditioning circuit of the signal. Note that most of the energy is concentrated in the area of 2 to 3 KHz.

This consideration has allowed us to change the sound card with a high-pass filter 3 KHz which produces many benefits including the minimization of the low frequency noise and the elimination of the omnipresent at 50 or 60Hz hum coming from the mains.



The filter high-pass, as well as eliminate the low-frequency noise, the impulse turns into "bipolar" (area of the yellow zone equal to the area of the green zone)

The bipolar pulses ensure a fast return to zero and the almost total elimination of sways the zero line. This facilitates the subsequent work of the "Baseline restorer"

Appendix 8 - bit ADC

For our "all software" approach 14 bits are the bare minimum.

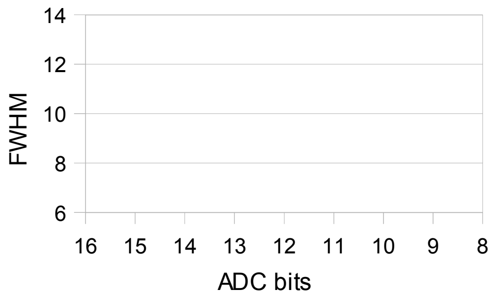
But 16 bits allow you to work comfortably, without the constant worry of saturation (if you amplify too) or the loss of dynamics (if you amplify a little)

With 12 bits the FWHM increases by more than a point and 10-bit down greatly increases.

We have verified the above statements by progressively eliminating the bits in software and seeing the resulting increase of FWHM. To repeat our test we must add the following line in the "DataArrived" module "WaveRecorder."

```
WaveInBuffer (i) = k * Cint (WaveInBuffer (i) / k)
// With k = 2,4,8,16,32,64,128,256 to degrade the ADC bit 15,14,13,12,11,10,9,8
```

ADC bits	FWHM
16	6.4
15	6.4
14	6.6
13	7.1
12	7.9
11	8.5
10	10.2
9	9.7
8	12.5



To eliminate any possible effect of this random testing was done always in the same conditions, with the same time and using several times the same data file (the file Cs-137 provided by Steven Sesselman and available on our website) Who repeated the test But keep in mind that values are "approximate" many factors (the parameters of the "baseline" for example) can also modify many points. The values at 14, 15, 16 bits, but the values change little with ADC to 11, 10, 9 and 8 bits are very dependent on the parameters. Note also that the 9-bit FWHM fell again, this indicates that intervened some different mechanism of error that might be eliminated by discarding the pulses relative.

Discarding most of the pulses (over 70%) the 8-bit value has dropped from 12.5% to about 10%. But even then discarding almost all pulses, and then slowing down terribly measures, they obtained no further improvement.

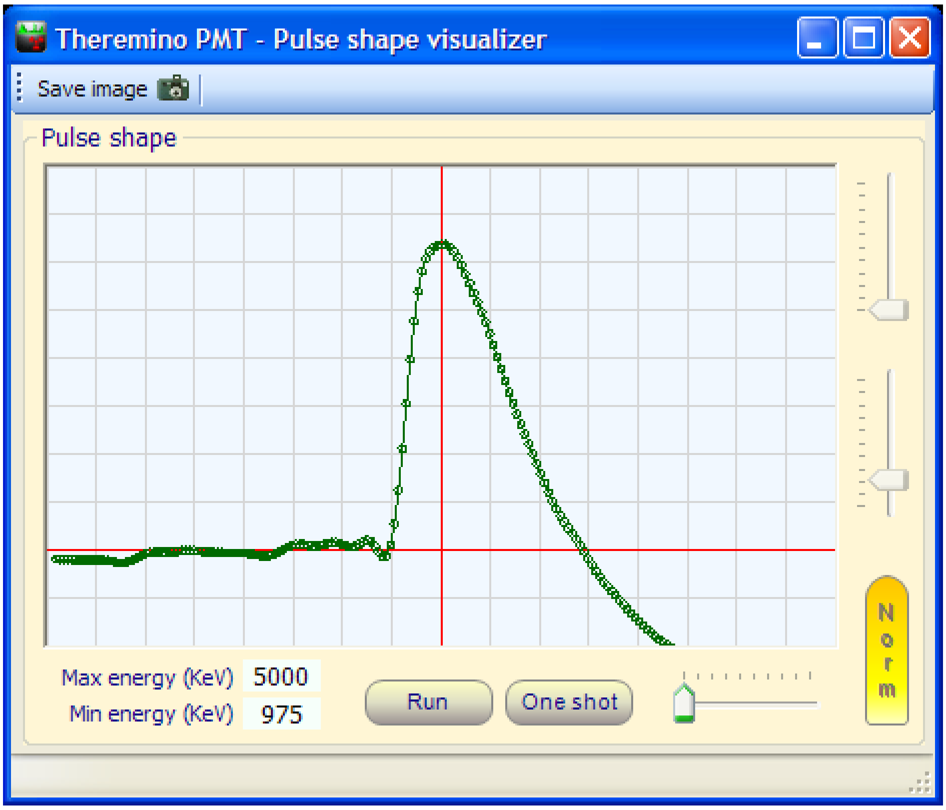
However, even if inaccurate and dependent on several factors, this test confirms our tests in the full of 2012, namely that the ADC must be at least 14 bits.

Some documents claim that may be enough 8-bit (with sliding scale), but in those conditions then you should do everything in hardware and use a sample-hold hardware. These same documents forget to write (or do not realize) that by doing so you are pretty much self-imposed limitations can not be resolved, they are unable to lower the FWHM.

Analog-to-digital components with steep edges, such as sliding-scale and sample-hold, are extremely difficult to bring together the delicate analog signal and **add a lot of noise,** probably also **ten times the noise that we consider too much.**

Appendix 9 - High sampling frequency

The sampling rate must be comparable to the width of the pulses, we use a frequency of 192 kHz with more than 100 uS wide pulse. The twenty samples per pulse that result allow to measure well, in addition to the pulse height, also other important characteristics, such as its shape, the amount of noise and the slope of the base line.



Note in this image the large number of samplings (green dots) in the vicinity of the tip.

If you do not stretch the pulses they would be a little fraction of uS and to get our performance you should use an ADC over 40 MHz and 40 MHz ADCs are difficult to use, consume a lot and have very few bits, 8 or 10 if okay.

And even if you could sample at 40MHz then you should be able to analyze continuously the more than 40 Mega Bytes of data per second that they come up. With normal PCs and embedded devices with much less this could not be done.

How do therefore hardware systems? They use a sample hold hardware and decide in hardware when it clicks. And so around the end of the work we do for algorithmic find "exactly" the tip of the pulse, and measure it as accurately as possible, it relies on two transistors that snap with little intelligence.

The hardware sampling methods are too rough, work well only when the pulses are spaced, large and clean, but as soon as there is noise taking a little earlier or a little later ... and these little errors that we do not make mistakes are precisely what allows us to scrape some fraction of FWHM more. And many villages of FWHM eventually make that whole percentage point that we were able to get.

Appendix 10 - From here on, all software

The analog and digital electronics is fascinating, but you must use it with greater sparingly as possible, it must convert the signal into digital immediately, less discrete components are welded and better.

The algorithms in hardware facts do what they can, by a narrow part from the limitation of complexity, from the other noise that revolves around it and without forgetting the leakage current, the parasitic capacitances, the ringing, the network at 50Hz, the errors design, the moist, trimmers scratch and the time that degrades them.

Making the algorithms in digital allows a completely different calculation precision and total absence of noise and is just so that we are able to recover many small-tenths of FWHM.

Our approach is only possible doing everything in software, with a CPU capable of doing floating point multiplications in a single clock cycle, which can be applied algorithms heavy buffer and digital filters on all samples, one by one.

Possible that you could do this with embedded systems, DSP, ATMEL, ARM or PIC (also Series 32) and even with mobile phones. They do not have the necessary power and do not have development environments suitable to the development of software as complex.

It took years to get these results while working with a development environment to a high standard and every facility possible. Doing the same things in a DSP, in a ATMEL, PIC or in a mobile phone would be like writing a book in the dark and with a tail of ostrich instead of the pen.

Anyone who thinks it is possible should try to do so, our software is all open source ...

Then there would be a small detail ... why?

1) The discourse on "the power added hardware" is not persuasive, rather it would be a miracle to be able to do everything that we do now.

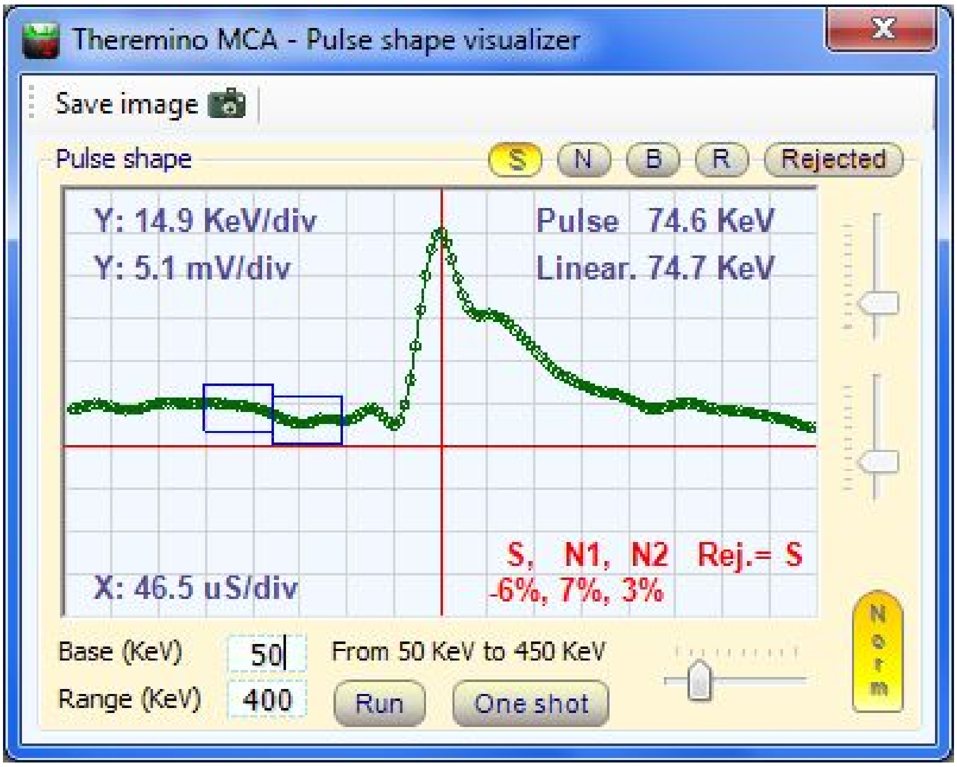
2) fast hardware, such as a DSP, would produce so much of that digital noise that our 200uV, hard-won, would no longer attract.

3) Add hardware, it would increase costs and difficulties of construction, without any benefit in return.

If the "advantage" that you want to get is a portable object then recommend a Netbook or Tablet or an EeeBox which moreover provide power, keyboard, and screen decently large and also batteries and chargers to CE standards. Without having to do everything by hand, only to find themselves with a shoe box full of wires and all in all, even less portable than a netbook.

If you should have any doubts about our claims Consider that over the past thirty years has been tried everything and no one fell far below 7%, however, with our approach can easily get to 6% and to 5%.

Appendix 11 - Precise measurement of the pulse



Try to measure the height of this impulse are 4.5 squares or 5 squares?

Hard to say it?

And enough errors **one twentieth of a picture** to add a **1% FWHM**

Higher is the precision with which we measure the height of the pulses and the lower the value of FWHM resulting.

Sometimes the presence of noise and crowds of many pulses neighbors can make it difficult to measure the height.

Even the presence of a line of zero swinging or the presence of ringing in the area which precedes the pulse can worsen the precision of measurement.

And also a sampling frequency of low (green dots little dense) increases the uncertainty of the measurements.

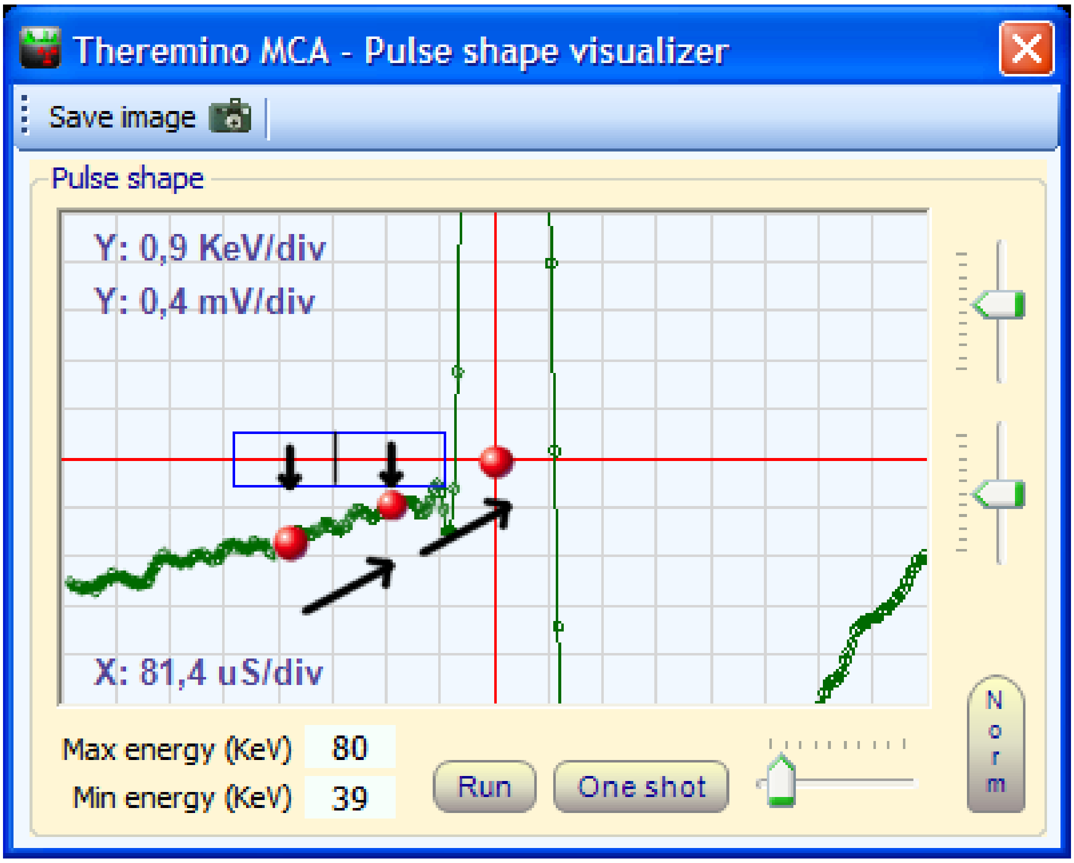
And the same happens if the bits of the ADC are few.

If the measurement error is small the pulse ends in a bin just next to the line and increases the value of FWHM. If the error is large (as in the cases in which is a mistake to identify the center of the pulse) the impulse ends up in a bin completely wrong and increases the noise floor.

In systems that work without a viewer of pulses and error the number of pulses measured imprecisely is much larger than you would normally think. Also we do not start we were unaware of it, after having hunted down one by one all the causes of problems, it has become apparent that systems "normal" number of pulses can be wrong even more than 10%. And this is one of the causes of the high noise floor which everyone considers normal.

Appendix 12 - Restoring The Baseline

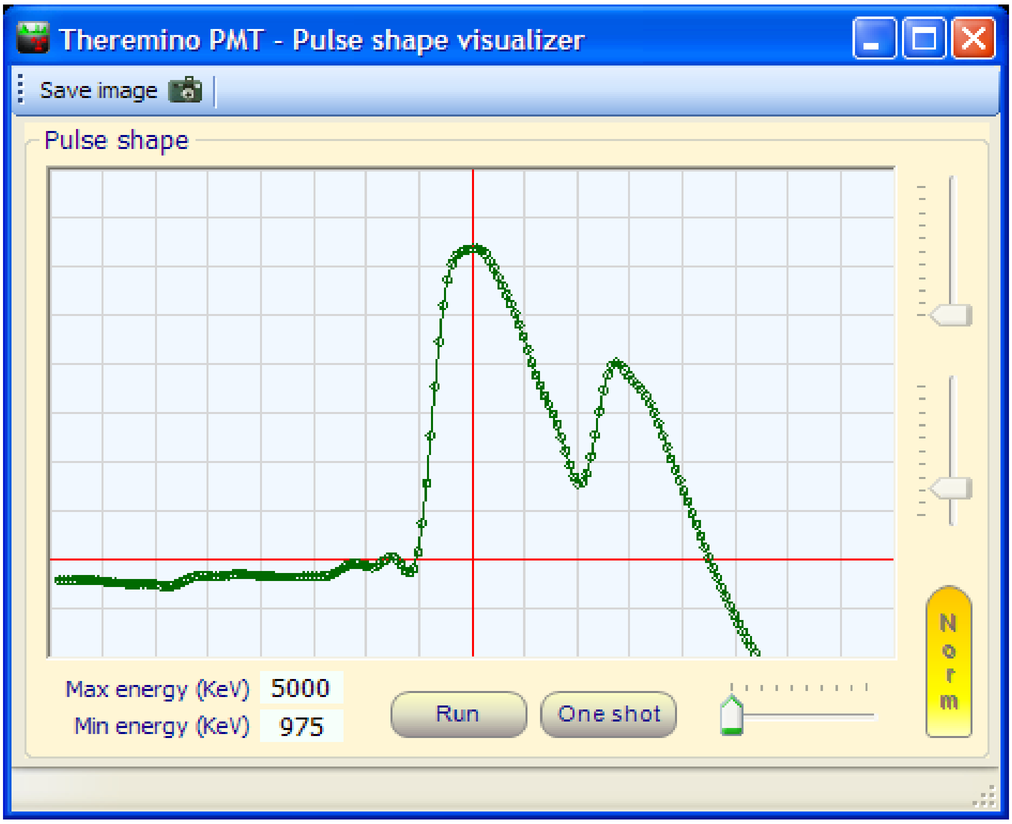
Offset compensation of the zero line with BaseLine Restoring.



The ThereminoMCA identifies the **zero position in the virtual center of the pulse** (The third red dot) with the highest possible accuracy.

Two averages are made to find the best estimate of the position of the first two red dots and finally a weighted extrapolation for the position of the third dot. This could not be done in hardware, and even if you did you would not have the necessary accuracy.

Appendix 13 - Identification of Pile Up



These two pulses are stacked, the second relies on the descent of the first.

You can measure the height of the first pulse with precision, but the second pulse is raised by the descent of the first and you can not find the location of its baseline.

So the first impulse is valid but not being able to measure the second it must be discarded.

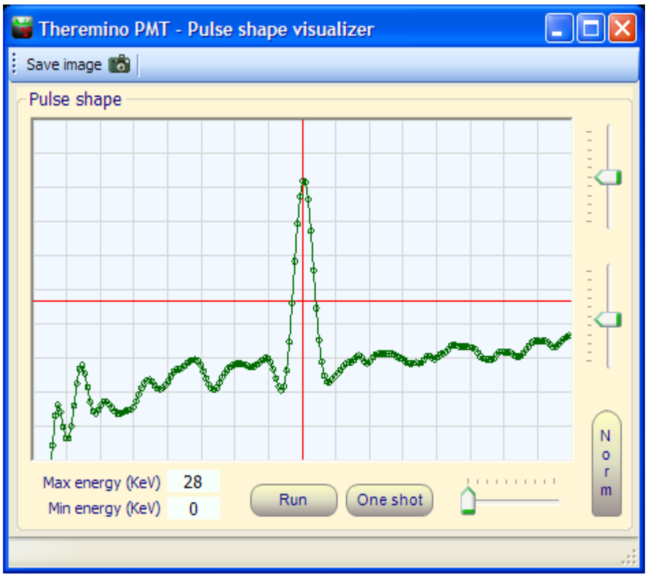
All PileUp must be identified in a reliable way, but when the situation becomes intricate due to the presence of noise or the overcrowding of many pulses neighbors, it can be difficult to discover them.

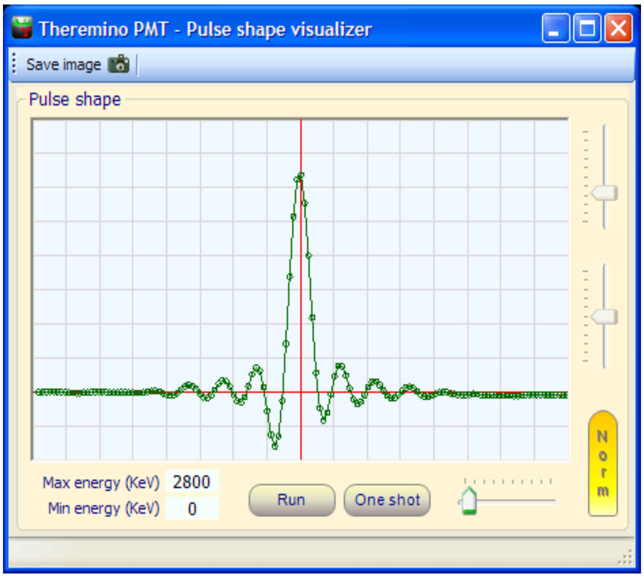
With the limited intelligence possible in hardware the number of PileUp not identified may be significant and each PileUp "escaped" generates a pulse measured bad, and each pulse measured evil widens the rows (if the error is small) or increases the noise floor (if the error is large)

Appendix 14 - Pulses too disturbed

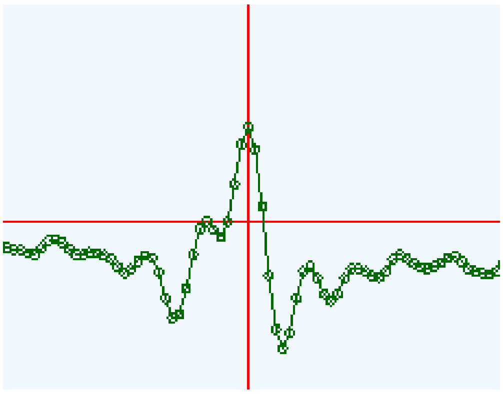
Identification of pulses too "noisy" (noise, ringing, or zero line pileup unstable)

To get the minimum FWHM all the pulses of the same height should end up in the same bin but every type of error can land them in a different bin from inaccurate measurement of the amplitude to the inaccurate evaluation of the zero.





Pulses with the zero line unstable Pulse affected by strong "ringing"

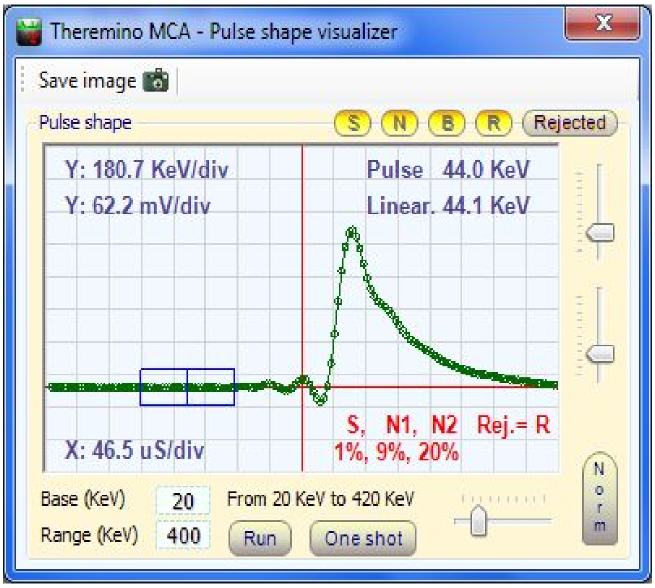


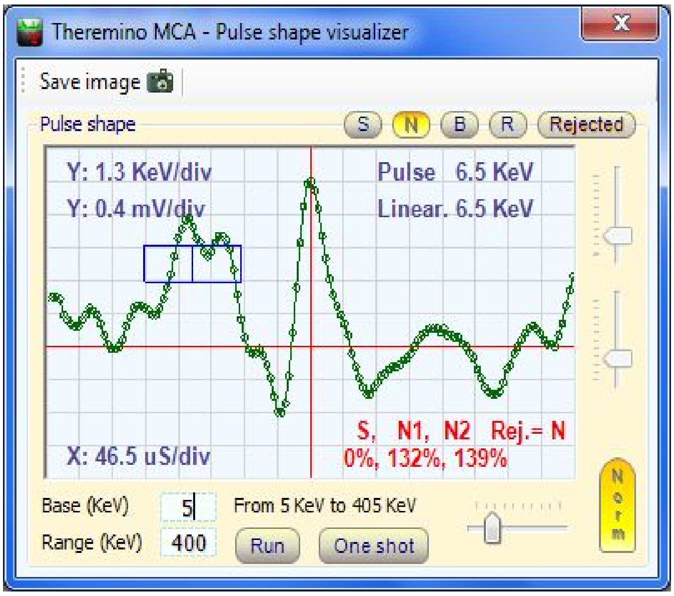
Pulse very disturbed

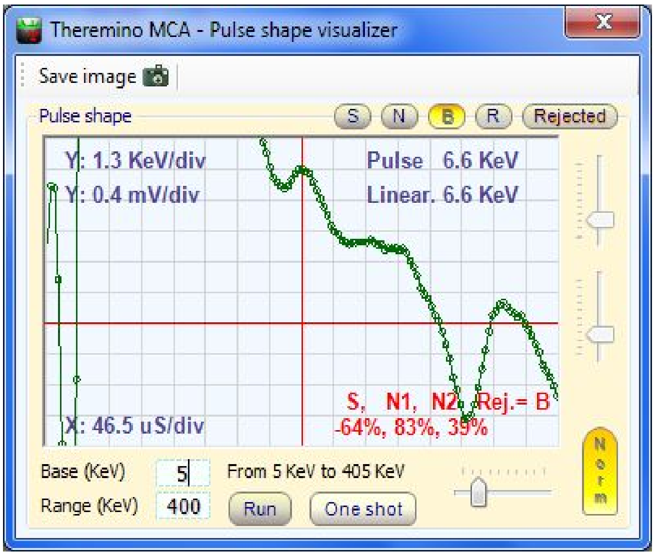
Some pulses, for various reasons, may be disturbed to the point of making it impossible to measure their height with precision. These pulses must all be recognized and eliminated.

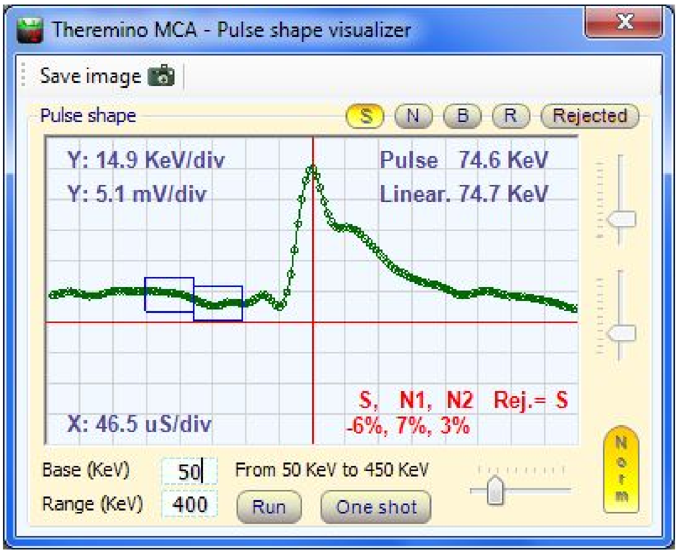
Each pulse measured evil widens the rows (if the error is small) or increases the noise floor (if the error is large)

Appendix 15 - Control functions









An instrumentation "service" able to detect errors and of showing the frequency and the type is essential to reduce the FWHM.

Also learn about the shape of the pulses and display cases "difficult" helps a lot to make a good "tuning" of the parameters and to identify any hardware defects.

There are a large quantity of strange events that occur in practice and that the theory forgets to consider.

Applications that do not have these tools are all in the dark and think they work well but it is not and their spectra show it.

Appendix 16 - Resolution compensation

Our "Resolution compensation" algorithm is a particular form of digital deconvolution.

DEFINITION: The deconvolution is a process based on algorithms used to reverse the effects of convolution on recorded data. The concept of deconvolution is widely used in the techniques of signal processing and image processing. These techniques are themselves widely used in many scientific and engineering disciplines, in which the deconvolution finds many applications. From: en.wikipedia.org/wiki/Deconvolution

The algorithm ThereminoMCA

According to the current literature, ours is a "deconvolution algorithm by derivation"

Before writing this document we did not know anything about deconvolution and now, after reading the literature, our algorithm is simple and naive. However, its results are comparable with the most elegant and mathematically correct algorithms, such as the document "Deconvolution of the Raman Spectrum of Amorphous Carbon" visible on the next page.

Our algorithm performs two scans of the spectrum, the first from bottom to top and the second from top to bottom (this can be corrected independently for the left and right side of the lines) During the past the value of each "BIN" is modified adding the difference (derivation) between its value and the value of a BIN which precedes it with a distance of a number of BIN (adjustable via the parameter "Size (bins)") and multiplied by a fraction of the parameter "Center (%) "The two additional parameters" Left (%) "and" Right (%) "are used to independently adjust the amount of correction on the left and right sides of the rows. Two other internal parameters (prefixed and not available to the user) regulate the amount of correction for the negative slopes than positive ones.

Thank you very much to the user "bepobalote" of the radioactivityforum which has strongly requested more formal informations about our methods, thanks to him we found the "canonical" versions of our algorithm (mainly that recursive) and **chances are good that applying them it will be possible to get another fraction of FWHM.**

The recursive deconvolution

One of the most well-known deconvolution techniques (called "recursive") involves several stages: calculation of the inverse Fourier transform of the spectrum, multiplication by a Gaussian or Lorentzian function and finally Fourier transform of the result. The process is repeated iteratively in the form of so-called tuning band.

The variable parameters of this algorithm are:

- The proportion of the form band: Gaussian or Lorentzian;

- The width of the band. If too small, the spectrum shows small variations in intensity, and if too large, it will produce bands negative laterally to those deconvoluted;

- The degree of intervention. If too small, the resulting spectrum does not differ much from the original, if too large, they produce false bands and the noise tends to increase.

An example of recursive deconvolution

A perfect example example of recursive deconvolution is in the document: "Deconvolution of the Raman Spectrum of Amorphous Carbon":

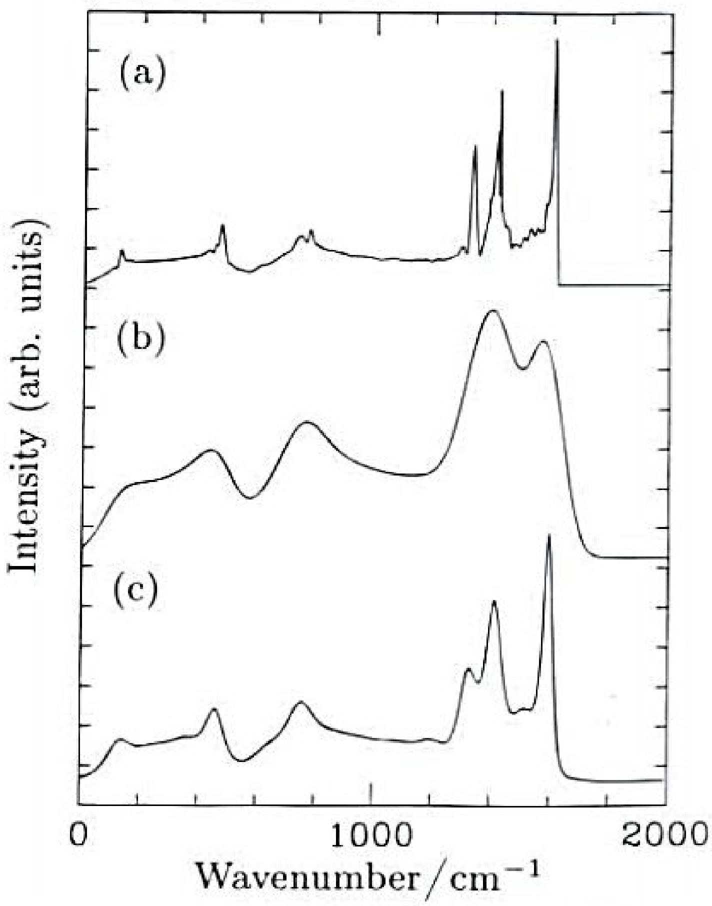


Figure 1. (a) The known spectrum taken as *o*. (b) The convolution of the spectrum in (a) with a Gaussian function with a width factor of 50 cm^{-1}. (c) The deconvolution of (b) using a Gaussian-shaped spread function with a width factor of 50 cm^{-1}. The iteration number is 200.

The Raman spectrometry produces spectra with very poor resolution (graph b) that surprisingly resemble those of Gamma Spectrometry with crystals Nai (Tl)

Panel (a) shows the spectrum that would be obtained with a detector perfect.

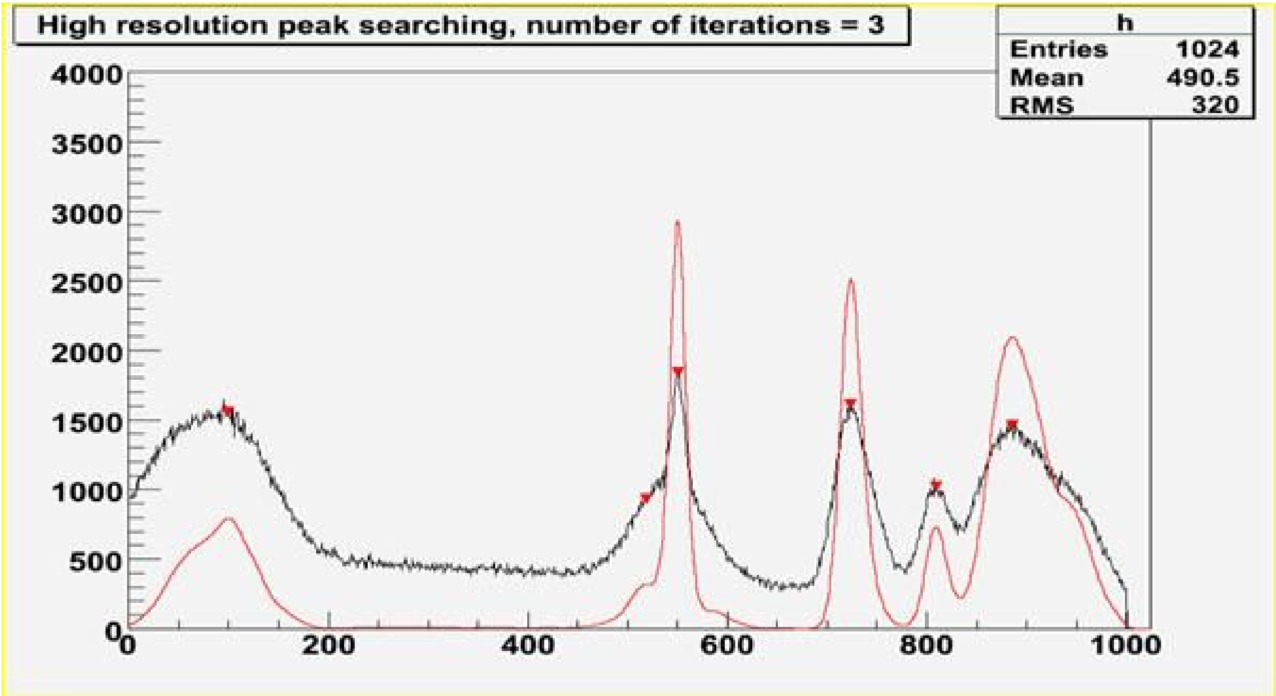
The graph (c) shows the increase of resolution which is achieved with the deconvolution.

They are practically the same results that we get, so similar to our spectra we could easily confuse them. And even the partial reconstruction of the original data and the improvement of FWHM appear very similar to those obtained with our algorithm.

An example of data from the Gamma Spectrometry

It seemed impossible that anybody had thought of it and in the end, look look, we also found someone who applies the deconvolution Gamma Spectrometry.

And who could it be but the CERN in Geneva, the magicians of the data collection? And who did the "dirty work"? A Slovak named Miroslav Morhac: **M.Morhac et al.: Efficient one-and two-dimensional Gold deconvolution and its application to gamma-ray spectra decomposition. Nuclear Instruments and Methods in Physics Research A 401 (1997) 385-408.**



For more information go here: http://root.cern.ch/root/htmldoc/TSpectrum.html

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It 's all open source and there are algorithms ready, just copy them. But it is said that in the end will transfer these classes in our software, for several reasons:

- Algorithms are heavy, slow, and recursive, intended to be applied only once and not ten times per second like we do.
- The adjustment of the parameters is cumbersome and less intuitive than ours.
- It does not appear that the improvements obtainable can justify the loss of which would cause interactivity with their slowness.

However, we will test them.

Limitations of deconvolution algorithms

If the data were impaired with a function, which we call F (x), which we might know, is the complete mathematical expression that its coefficients, it is sufficient to apply the inverse function to get exactly the original spectrum.

The problem of knowing the coefficients can be solved experimentally by finding the values that give the best results and the function is generally a Gaussian with slight deformation caused by certain mechanisms minors.

Unfortunately there are also reasons that prevent a thorough knowledge of both the function of its parameters.

When a gamma ray disintegrates in the crystal generates a large number of photons. A second point of the crystal in which this conversion is the number of photons generated is slightly different (due to the non-perfect homogeneity of the material) is always depending on the position the number of reflections that the photons must perform against the walls before reaching the photomultiplier changes. To which generates an uncertainty in the amplitude of the generated pulse.

You should therefore **apply a different function for each pulse**, with slightly different parameters and you should do this **on the basis of knowledge that we can not have**.

Fortunately the sum of all errors results in a roughly Gaussian function with the overlap of some random variations. The gaussian may be correct, but no random variations. **Therefore, the reconstruction will be only partial.**

If you overdo it with the amount of correction are generated artifacts. And further increasing the correction artifacts increase until the resulting curve becomes worse than it was intended to correct.

Objections

Does anyone think that the techniques of deconvolution can not add anything to the raw data and says "what is subtracted is now lost and you can not recreate." But since you know some characteristics of the function that has degraded the data, just "recreate" the missing parts "adding" the knowledge we have to get a real and effective improvement of data quality.

Conclusions

The possibility exists to improve, at least in part, the data of the spectrum.

The correction is not just cosmetic or apparent but actually increases the resolution of the probe.

There is a certain set of parameters that guarantees the highest quality of correction, and these parameters can be found experimentally.