Using Adaptive Equalization in Modulation Analysis and Troubleshooting

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Many digital wireless systems use adaptive equalization to improve transmission range, reliability, and carrying capacity. The equalization accomplishes this by dynamically creating and applying a compensating filter, removing linear errors from modulated signals. These errors include group delay distortion, frequency response errors (tilt, ripple), and reflections or multipath distortion.

Analysis and troubleshooting of such systems can be made more efficient by using this same technique in measurement tools. Indeed, some impaired signals can only be measured *after* equalization.

This paper describes adaptive equalization and its benefits in modulation analysis and troubleshooting.

The purpose of test equipment in wireless design and manufacture is twofold: to understand and predict the performance of a system and to find and quantify problems so that they can be fixed (or ignored, if appropriate). To do this, test equipment is often called upon to mimic all or portions of the functions of a transmitter or receiver.

To test complete transmitters or to test components such as modulators and power amplifiers it is desirable to have some sort of ideal or perfect receiver. Test equipment has historically been called upon to perform this function, and it has been a constant effort to keep such equipment up-to-date with the components and systems it is designed to test.

Nowhere is this more true than in the field of digital communications. Several factors conspire to hamper the development of the ideal measuring receiver or demodulator. They include the fast pace of design and innovations, the complexity of systems and measurements, and the highly variable nature of the specifics of these systems.

Adaptive equalization is a good example. Though the technique is used in many digital radio receivers, it is only now starting to appear in the test equipment normally used to design and test these receivers. Since receivers which use adaptive equalization already exist commercially, it is obviously possible to design and manufacture them without test equipment so equipped. But it is considerably more difficult.

Fig. 1 illustrates one form of the difficulty. Does this measurement represent an acceptable 16QAM signal or an unacceptable one?

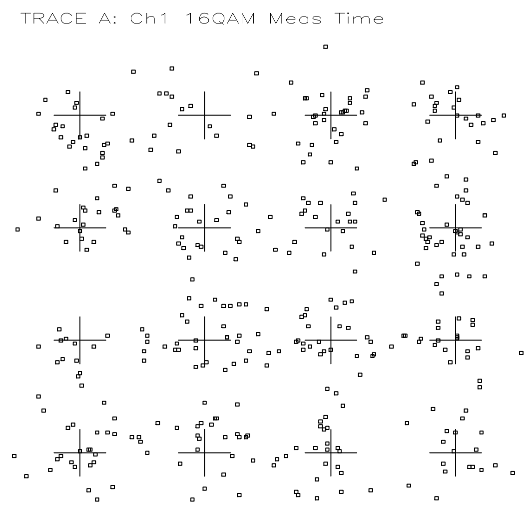

Fig. 1. 16QAM signal without equalization--15% EVM

The same question can be asked about the measurement in Fig. 2.

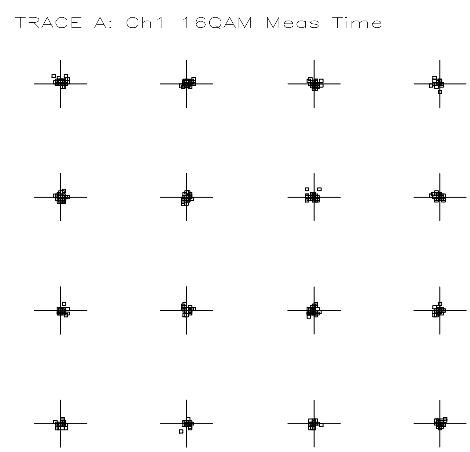

Fig. 2. 16QAM signal with equalization--1.5% EVM

Of course, these two measurements are of the same signal. The difference is the presence of an equalization filter in the measuring receiver, which reduces the error in the signal by an order of magnitude. Whether the signal quality is deemed acceptable or not is a decision that can now be made with full knowledge of how a real receiver (with adaptive equalization) would respond. In a less-extreme situation this kind of knowledge may allow an otherwise good modulator or transmitter to be used instead of thrown away.

Adaptive Equalization Primer

Since this paper covers the use of adaptive equalization rather than the theory, only a brief summary of adaptive equalization itself will be provided.

Equalization is the process of applying a filter to a signal to remove or compensate for the effects of linear distortion. This filter can be defined in the frequency domain by frequency response parameters such as gain and phase or group delay. Alternatively it can be defined in the time domain by its impulse response.

Adaptive equalization dynamically creates and applies such a compensating filter, modifying it to more completely compensate for distortion or to track changing signal characteristics.

Equalization is effective at compensating for linear distortion mechanisms including:

- Frequency response errors such as ripple or tilt
- Non-flat group delay or nonlinear phase
- Multipath or delay spread

However equalization in not effective on nonlinear distortion mechanisms such as:

- Noise
- Spectral regrowth or adjacent ch. interference
- Intermodulation
- Spurious
- Harmonic distortion

This lack of effectiveness on nonlinear distortion mechanisms in an unfortunate fact from the standpoint of receiver performance. However if the equalizer capability can be made switchable in the measuring receiver (test equipment), the result is an effective way to distinguish linear from nonlinear distortion. This has considerable implications in the testing of modulators, transmitters and transmission paths. These implications will be covered later in this paper.

Equalizer Types

Equalizers can be sorted into two main types: *Feed-forward* and *decision feedback*. A diagram of the feed-forward equalizer is shown in Fig. 3.

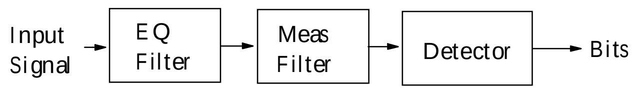

Fig. 3. Feed-forward equalizer topology

In this equalizer the input signal is passed through the equalization filter and then through any other filtering required by the system. Following detection or demodulation the signal is measured. The result of this measurement can then be used to refine the equalization filter.

The decision feedback equalizer topology is shown below in fig. 4.

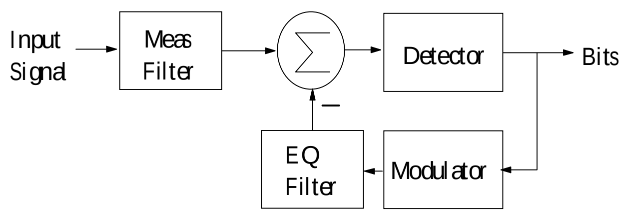

Fig. 4. Decision feedback equalizer topology

In this equalizer the demodulation result (recovered bits) is used as the input to a modulator circuit. The output of this modulator (an ideal, noise-free signal) is then filtered and subtracted from the input signal. This allows equalization to be performed without the equalization process itself adding noise to the signal.

Equalizer Algorithms and Training

All adaptive equalizers require some mechanism for adaptively changing filter parameters or "training" the equalizer. Equalizer training can be sorted into two main types: *Decision-directed* and those that use *training sequences*. Decision-directed training is illustrated in Fig. 5. The LMS (least mean square) algorithm is described in [1].

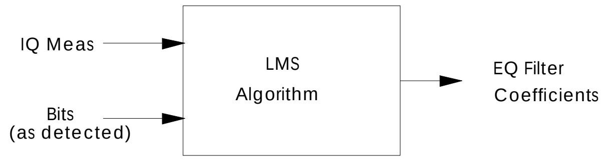

Fig. 5. Decision-directed equalizer training

Decision-directed equalizer training requires no prior knowledge of the transmitted data. This can be an advantage in test equipment where the necessity to provide such a sequence (and its exact position in a transmission, etc.) is avoided. However the training action (filter convergence) is slower and less certain when there are bit errors and is then often too slow to follow rapidly-changing channel characteristics.

For these reasons, an alternate method of training which uses a training sequence is more common in practical radios. It is shown in Fig. 6.

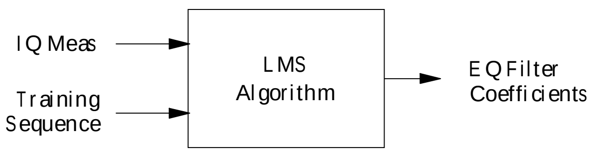

Fig. 6. Equalizer training using a training sequence

The use of a training sequence implies prior knowledge of a portion of the transmitted data. The training sequence carries no information and therefore reduces the payload of the system. However the sequence allows the equalizer to converge in the presence of many bit errors and can allow the equalizer to converge fast enough to follow rapid changes in the channel.

Adaptive Eq. in Test and Troubleshooting

Adaptive equalization in a flexible test instrument provides a variety of benefits. They include:

- Evaluating a signal in the same way that a receiver (equipped with adaptive equalization) would.
- Ability to separate linear from nonlinear distortion mechanisms so that design and troubleshooting effort can be concentrated on the important ones.
- Ability to measure small nonlinear distortions in the presence of large linear distortion that would normally obscure them.
- Ability to measure impaired signals. Accurate measurement of some impaired signals may require equalization to obtain a symbol lock and successful demodulation.
- Ability to derive important parameters of the linear distortion such as the frequency response of the channel or the timing of delay spread.

To demonstrate these benefits we will make use of measurements and displays from the HP 89441A Vector Signal Analyzer, the first general-purpose signal analyzer to offer an adaptive equalization capability.

Comparable Measurements

In systems where the receivers include equalization, many linear errors can be compensated for, and do not cause loss of information. Therefore measurements made in the lab or on the production line to evaluate the actual performance of a system should also include equalization.

If equalization is not performed it may be very difficult to predict the actual performance of a system. In addition, good transmitters or modulators may be rejected and much time or money may be spent in optimizing a design or adjusting a modulator or amplifier to minimize a source of distortion which has no real consequence.

In such measurements the equalizer in the test equipment must perform the same general function as the receiver equalizer, but an exact match of equalizer topologies or training techniques is not required.

Linear vs. Nonlinear Distortion

The ability to isolate problems and error sources is vital to successful system integration and troubleshooting. As mentioned previously, equalization is able to reduce the effect of linear error mechanisms only. By switching the equalization on and off in the precision receiver, the magnitude of both distortion types can be measured.

Of course the nature (linear or nonlinear) of a distortion source is a clue to its origin and to potential remedies. Fig. 7 below shows the constellation of an 8PSK signal without adaptive equalization applied. Measured EVM is approximately 8%.

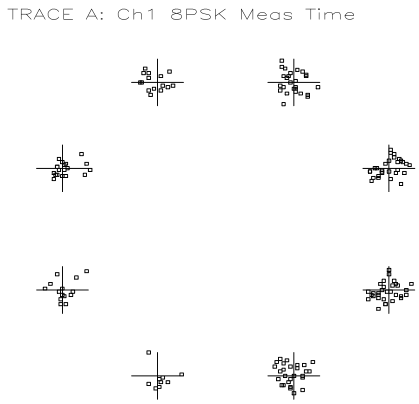

Fig. 7. 8PSK measured constellation without equalization

With an adaptive equalizer in the receiver the EVM is reduced to 5% as shown in Fig. 8 below.

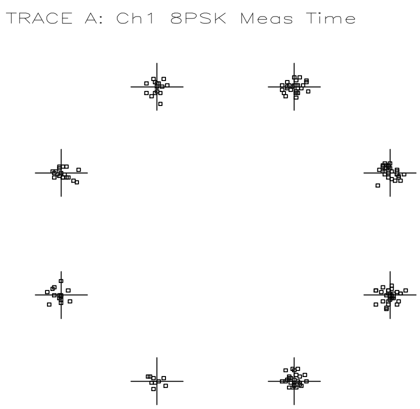

Fig. 8. 8PSK measured constellation with equalization

Other measurements (in addition to EVM) can now be made on the equalized signal to determine the nature of the nonlinear distortion. A flat EVM spectrum for example, would indicate a noise problem. An EVM spectrum raised at a band edge would indicate adjacent channel interference.

Measurements such as these could also be used to establish and manage a system error budget with separate categories for linear and nonlinear mechanisms, along with the usual categories for different system blocks.

Uncovering Smaller Distortions

In systems where receivers are equipped with equalizers it may be important to find and eliminate nonlinear distortion sources, even when their error contribution is much smaller than that of linear distortion sources.

Equalized error measurements are an obvious answer to this need. Consider the QPSK constellation in Fig. 9.

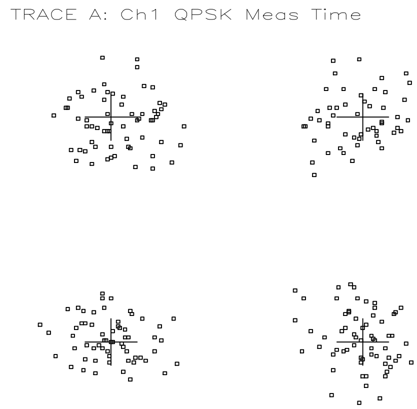

Fig. 9. QPSK measured constellation without equalization

This signal has both linear distortion and a small spurious signal which is in-band. That is not evident, however, from either the constellation display in Fig. 9 or the error vector spectrum measurement in Fig. 10 below.

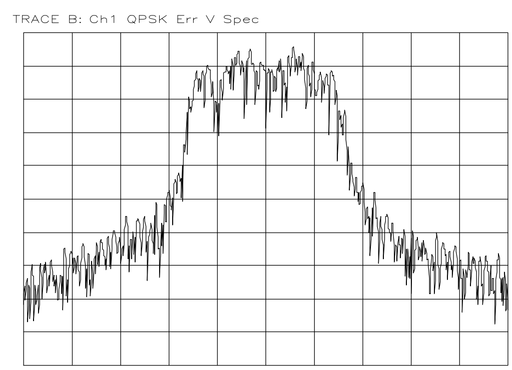

Fig. 10. QPSK meas. EVM spectrum without equalization

Adding an equalizer to the precision receiver removes the linear distortion and significantly changes both the measured constellation and EVM spectrum.

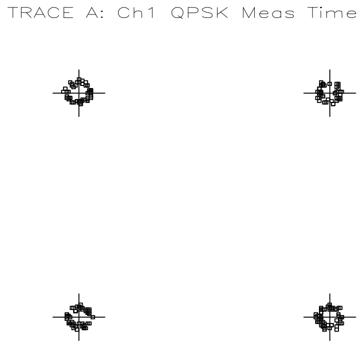

Fig. 11. QPSK measured constellation with equalization

The constellation in Fig. 11 clearly shows the effects of spurious interference. This is confirmed in Fig. 12 below.

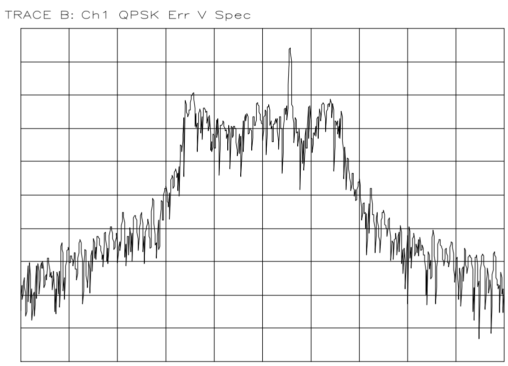

Fig. 12. QPSK measured EVM spectrum with equalization

The broadband error energy in the measurement has been considerably reduced, revealing a spurious signal in the channel. Its frequency can now be accurately measured, providing a clue as to its source.

Measuring Impaired Signals

In real-world situations some signals are impaired by large amounts of linear distortion. This is particularly true of over-the-air transmissions where multipath or delay spread is a problem. Without equalization in the receiver, these impairments may be too severe to allow symbol clock recovery and successful demodulation. An example is shown in Fig. 13.

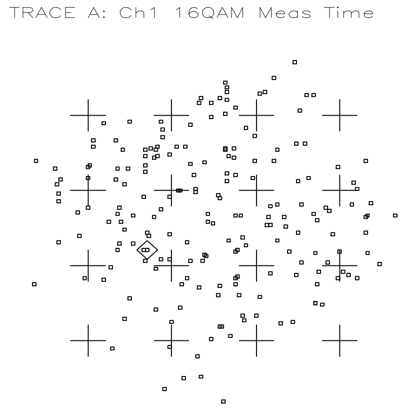

Fig. 13. 16QAM signal before equalization

It may be suspected from the dots in the constellation that this is some sort of QAM signal with a square constellation. However the receiver is unable to regenerate a symbol clock and no quantitative information can be gained from this measurement.

Adding a decision-directed (no training sequence) adaptive equalizer to the circuit quickly improves the measurement. Fig. 14 below shows the early results while the equalizer is being trained.

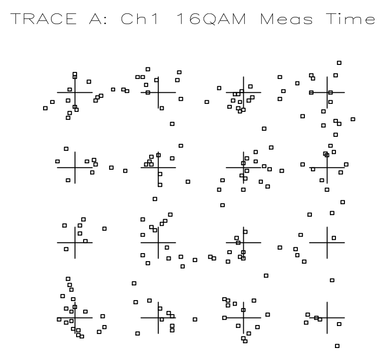

Fig. 14. 16QAM signal after 8 measurement updates

The symbol clock in the receiver is clearly now in a locked condition. Symbol errors are still present but demodulation is now successful.

Once symbol lock is achieved the equalizer trains (converges) much faster. Fig. 15 shows the measurement result after just 4 more measurement updates.

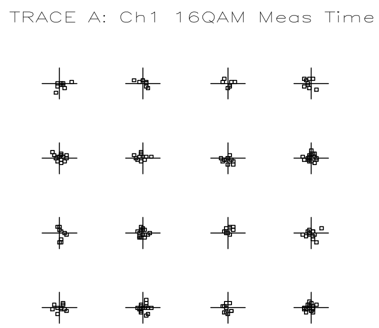

Fig. 15. 16QAM signal after 12 measurement updates

Measured EVM has been now been reduced to just 3% on a signal which could not be demodulated at all before equalization. The ability of this general-purpose adaptive equalizer to successfully train itself without a training sequence and without initial symbol lock is obviously important for severely-impaired signals.

Deriving Distortion Parameters

Once trained, the receiver's adaptive equalizer filter is itself an important source of information. In the HP 89441A, its complex filter coefficients can be viewed in the form of the impulse response of the equalizer filter. They can also be represented (in inverse form) as the imputed frequency response of the channel.

Figure 16 below is an example of an equalizer filter impulse response.

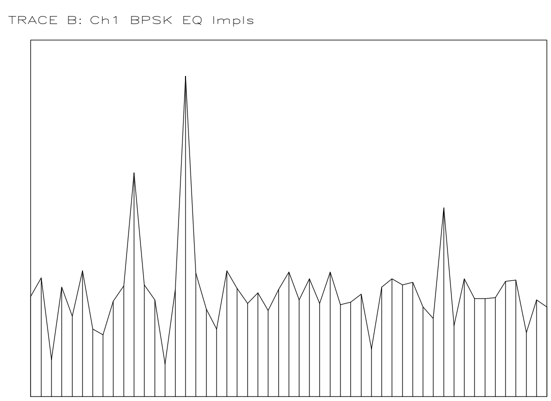

Fig. 16. Equalizer filter impulse response

The filter tap magnitudes are shown on the Y-axis in log magnitude format while the X-axis is linear in time or in symbols. In this case the symbol times are shown by the vertical bars.

A display such as this can be useful in determining the relative magnitude and timing of multipath or delay spread. The largest response represents the strongest signal received and the smaller responses represent weaker alternate paths. Note that one of the weaker signals has a negative delay relative to the strongest signal. This would indicate a situation where the direct (shortest) path has more attenuation than one of the alternates.

To understand linear distortion in a transmitter it is more useful to look at frequency response parameters such as gain and phase or group delay. The frequency response of a channel can be derived from the impulse response of the equalizing filter as shown in Fig. 17. This is a filtered version of a signal from an NADC (IS-54) transmitter.

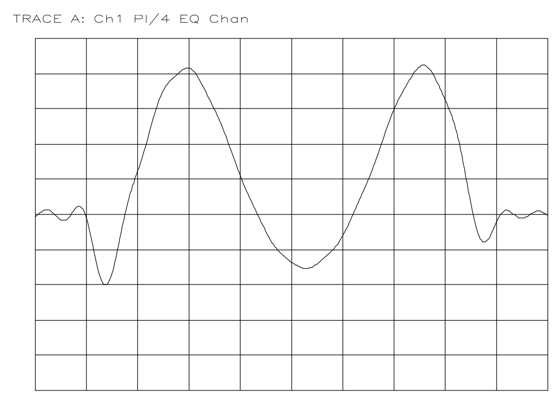

Fig. 17 Channel frequency response, log magnitude

This measurement of the channel frequency response is in log magnitude format and peak-peak ripple is approximately 2.8 dB.

Since the filter coefficients are complex, the derived channel frequency response can also be expressed as phase or group delay.

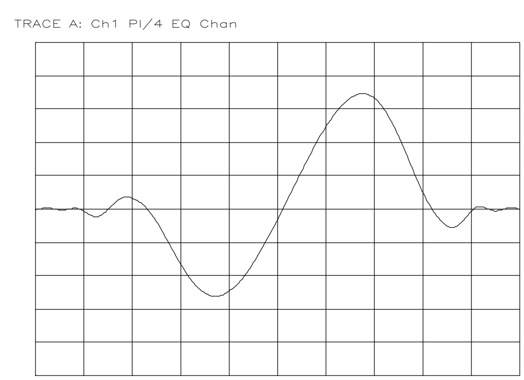

Fig. 18 Channel frequency response, phase

Fig. 18 above is the phase of the channel frequency response. Peak-peak phase variation across this filtered channel is 30 degrees.

Fig. 19 below shows the group delay or phase linearity of the channel. This filtered channel deviates from ideal (constant) delay by approximately 20 microseconds.

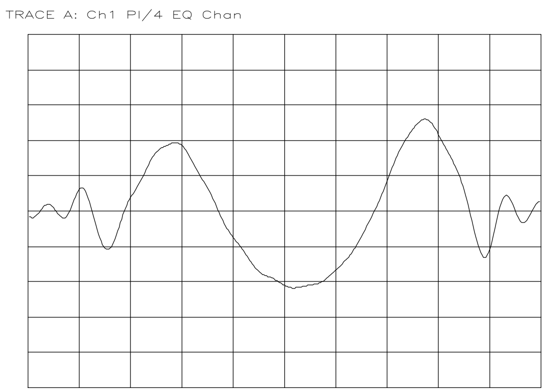

Fig. 19 Channel frequency response, group delay

The information generated by, and derived from adaptive equalization is not a universal substitute for traditional network analysis. However it is an accurate tool for troubleshooting and system verification. It has the important added advantage of using the (NADC in this case) signal itself as a network stimulus. Therefore it can be used as a single-ended, in-service measure of transmitter characteristics and/or channel impairments.

Restrictions and Limitations

As demonstrated here, adaptive equalization implemented in a general-purpose, precision receiver has several important uses. However any generalized implementation will also have some limitations relative to implementations which are application-specific.

This adaptive equalizer does not operate in *real time*. It is a block-mode implementation and does not continuously follow the input signal. Some portions of the input signal are usually not measured because the analyzer does not acquire new data while it is processing previous data.

Since it is not real time and is deprived of the information in a training sequence, this equalization adapts more slowly and is therefore restricted to "stationary" signals. For example, this rules out its use for in-service measurements in moving vehicles. Fortunately a time capture mode with extensive overlap processing is available in the HP 89441A and this can be used in some cases to make signals appear to be stationary.

This general-purpose equalizer is not usually as robust as those designed for a specific application. Their parameters can be carefully selected to optimize convergence and stability and they often benefit from training sequences.

For similar reasons, the results produced by this generalized receiver will not exactly match those from an application-specific receiver. Differences in equalization and training techniques and in operating parameters will yield some differences in demodulation results.

Finally, this equalization algorithm cannot be guaranteed to be stable under all conditions. Equalizer training is similar to the operation of a control system, and under some conditions the training algorithms will diverge or fail to converge.

Despite these limitations, this general-purpose implementation of adaptive equalization has significant benefits for both the design and manufacturing of wireless systems. This is true whether or not the systems to be measured use adaptive equalization themselves.

References

[1] "Block Implementation of Adaptive Digital Filters," IEEE Transactions on Circuits and Systems, June 1981, Vol. CAS-28, No. 6.