December 2004
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ANODE

Inside this issue:

Special points of interest:

- Contact details on back page (updated)
- New email address for Anode and ZS6WR. See back page

Editor's Comments

January 2005
Volume 5 Issue 06
Things I have found this year so far:

Demo programs that still don't comply with Y2K or work properly in 2005.

Alkaline batteries that leak.

Experilab at Princess Crossing has been running courses for young experimenters during the school holidays. They have decided to run a course on pic's & programmers in the evening. Contact Experilab on 011 768 7870 for more details or email shop@e-lab.co.za

I have not yet received a 2005 Calendar from the club, so nothing can go wrong this year.

Windows XP Service pack 2 = 'Mostly Harmless'

[For Bob ZS6RZ]
What are Gray codes, and why are they used?

The correct spelling is "Gray" - not "gray", "Grey", or "grey" - since Gray codes are named after the Frank Gray who patented their use for shaft encoders in 1953 [1].

Gray codes actually have a longer history, and the inquisitive

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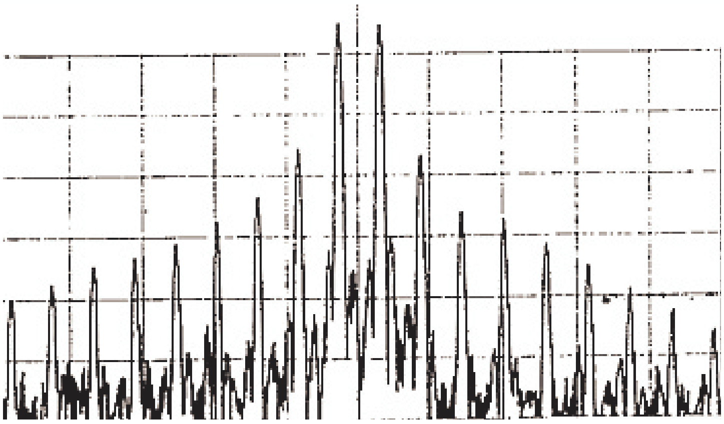
Intermodulation Performance and Measurement of Intermodulation Components

by Lloyd Butler VK5BR



Introduction

To define the performance of a receiver or transmitter, various specifications are recorded which are obtained from measurements carried out. Perhaps the least understood of these in amateur radio circles is intermodulation performance and how it is measured. The aim of this article is first to discuss intermodulation products and how they are produced and then look at how they are defined and measured.

What are Intermodulation Products?

When a single frequency (f1) is fed

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Editors Comments & News

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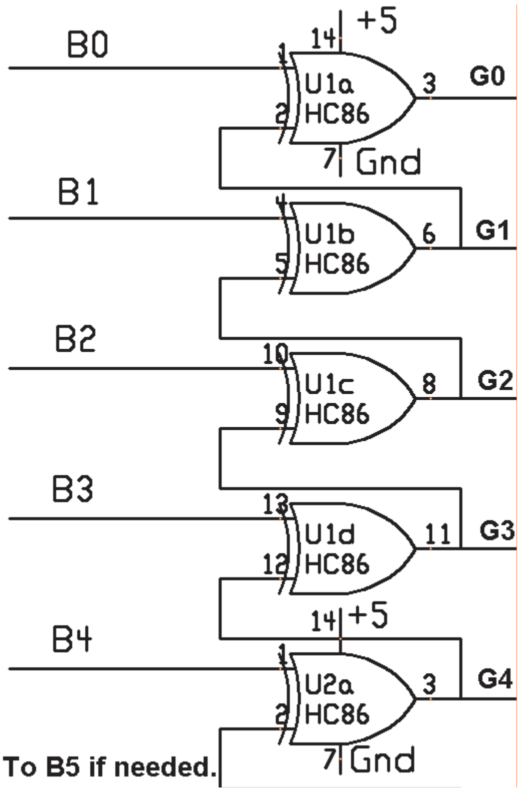
reader may want to look up the August, 1972, issue of Scientific American, which contains two articles of interest: one on the origin of binary codes [2], and another by Martin Gardner on some entertaining aspects of Gray codes [3]. Other references containing descriptions of Gray codes and more modern, non-GA, applications include the second edition of Numerical Recipes [4], Horowitz and Hill [5], Kozen [6], and Reingold [7].

A Gray code represents each number in the sequence of integers {0...2^N-1} as a binary string of length N in an order such that adjacent integers have Gray code representations that differ in only one bit position. Marching through the integer sequence therefore requires flipping just one bit at a time. Some call this defining property of Gray codes the "adjacency property" [8].

Example (N=3): The binary coding of {0...7} is {000, 001, 010, 011, 100, 101, 110, 111}, while one Gray coding is {000, 001, 011, 010, 110, 111, 101, 100}. In essence, a Gray code takes a binary sequence and shuffles it to form some new sequence with the adjacency property. There exist, therefore, multiple Gray codings for any given N. The example shown here belongs to a class of Gray codes that goes by the fancy name "binary-reflected Gray codes". These are the most commonly seen Gray codes, and one simple scheme for generation such a Gray code sequence says, "start with all bits zero and successively flip the right-most bit that produces a new string."

Decimal	Binary	Reflected Binary	Hexadecimal
0	0000	0000	0
1	0001	0001	1
2	0010	0011	2
3	0011	0010	3
4	0100	0110	4
5	0101	0111	5
6	0110	0101	6
7	0111	0100	7
8	1000	1100	8
9	1001	1101	9
10	1010	1111	A
11	1011	1110	B
12	1100	1010	C
13	1101	1011	D
14	1110	1001	E
15	1111	1000	F



The table shows the various values. GRAY code is otherwise known as 'reflected binary'

The circuit shows how to translate the binary (B0 to B4) to GRAY code (G0 to G4). 74HC86|7486 or any exclusive OR gate should do the job.

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Beeswax in an oscillator compartment

From: "Larry Gagnon" <lagagnon@fakeuniserve.com>
Subject: beeswax in an oscillator compartment
Date: 2004-12-15 23:21

I posted a few days ago about repairing a VHF marine radio. Thankfully I solved my problem, locating three cold solder joints in the VCO, which entailed removing metal shields and all the beeswax that coated all the components. That was character building work!

I gather the beeswax is to ensure rigid components to maintain stability of the oscillator? Any other reasons for it? Is it necessary for me to remelt the old wax and spread it around as before, or can I get away with just re-shielding the VCO?

Any suggestions appreciated.

Larry VE7EA

[to reply via email remove "fake" Microsoft will soon release their newest product: a vacuum cleaner. It will be their only product which doesn't suck. Hi,]

Go ahead and remelt it. We always did when I used to work in Midland's Service department back in the 70's. Never caused a problem.

Beeswax has been used for securing the windings on toroids.

According to my boss in the late '60s and early '70s,W.T.G. Glasspool (would you guess he was British?), beeswax was widely used in the past to secure coil windings, seal slugs in the cores of forms and transformers, etc.. Apparently it doesn't reduce the "Q" too much, and was readily available in those days.

73, John - K6QQ

I purchased some here in the UK from a local craft shop quite recently. It is an excellent material for holding things together as long as the temperature doesn't get too high and IMHO makes a neater job than hot-melt for holding small items onto a PCB. The trick is to keep an old iron bit especially for the job (and also for starting holes in plastic boxes.) The stuff I got is in the form of small beads and so very easily handled with a pin. The wax in that VCO, BTW, is anti-microphonic in purpose and definitely should be re-melted. This is particularly important if the loudspeaker is in the same cabinet with the radio. Cheers - Joe, G3LLV

Beeswax goes back the the 1930s USA designed radios too.

In common radio design use since then. >It is excellent as it has a high melting point (for waxes).

Just put some it back on, it keeps the coils/wires stationary.

You should only need a line on both sides of the coil.

Quite true on old-time radio production, but primarily for the lower-cost "consumer" type models.

The wax isn't from bees, but rather from other sources and is usually called "ceresin wax." Pours easily when hot, stays hot enough for a quick brushing-on.

Unfortunately, hot spots in old tube/valve equipment lets the wax soften and it sometimes dribbles off and quits holding what it was supposed to hold.

O-T story: Back in 1956 when I was new to WREX-TV and on midnight maintenance shift, I was supposed to align the air monitor scope that sampled the transmitter output signal. The video response was way off judging by the sync signal appearance. Cause was the peaking coils of the internal tube video amplifier. An hour spent with a video sweep generator and a hot iron brought the peaking coils (pie-wound inductors sliding on 1 Watt resistor bodies) into a good, flat video response. Early day shift complained long and loud about "the air monitor doesn't work!" It showed the correct waveform, not the one they

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were used to! :-)
[nobody had been able to fix it before and all it took was to move the coils slightly to adjust their inductance]

There's nothing special about the wax. It was a convenience to use in production, quick, easy-to-use and cheap. To do a good hold on things like inductors, I would recommend the "spar varnish" kind of varnish (made from petroleum distillates, not the urethane type). One example of that is McCloskey "Gym Seal" intended for hard-use floor finishing and found in lumber yards and do-it-yourself stores. Varnish needs to cure overnight to be effective, doesn't drop Q any more than the polystyrene "Q Dope" (which will lose its holding power because of moisture breaking the bond to the work), and holds on infinitum (also known as at least 30 years based on my experience).
LenAnderson@ieee.org

Actually, the beeswax is a natural by-product of 30 hertz oscillators. Always wear gloves before opening!
:)
The Eternal Squire

Beeswax goes back the 1930s USA designed radios too. In common radio design use since then. It is excellent as it has a high melting point (for waxes). Just put some it back on, it keeps the coils/wires stationary. You should only need a line on both sides of the coil.

Remember "fish-paper"? used that in a early 1980 moto radio design.

I know the type of wax you are talking about, harder and a little darker than beeswax.
"Fish paper" (not made from fish) was common in E-I lamination transformer construction as a winding insulator. Actually it IS paper but impregnated with a polymer, good for tight wrapping of wire layers because it does not have the "give' of many plastic film wraps. Thin cloth impregnated with phenolic resin was sometimes used for cheap insulation as a substitute for "fish paper" on non-transformer structures.

If someone is intent on using any wax, then paraffin wax would be a bit better since its melting temperature is higher than ceresin waxes. Paraffin blocks can be found in larger do-it-yourself stores such as Lowe's or Home Depot. Harder to work with than the low temp-melting ceresin wax for potting small things.

LenAnderson@ieee.org

Beeswax and encapsulation seemed to be the fad for the Japanese mfg. in the early 1980s. I have heard various reasons, but mobile operation (with potential bouncing and jarring) is often mentioned. Kenwood amateur gear had some noted PLL problems with their encapsulation materials of that period -- although they were not the only mfg. using this method.

I am curious to know the age of the transceiver and mfg.

Greg w9gb

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Direct conversion spectrum analyser

From: "Avery Fineman" <averyfineman@aol.com>
Subject: Re: Direct conversion spectrum analyser
Date: 2003-10-14 23:16

In article <bmgam6$mgi8n$1@ID-61331.news.uni-berlin.de>, "Hans Summers" <Hans.Summers@Tudor.Com writes:

"Ashhar Farhan"
<farhan@hotfoon.com> wrote
in
mesage news:8331b126.0310132158.4a53459c@posting.google.com...

here is a spectrum analyser design that i would like the group to comment upon.

1) we take the input via a low pass filter, up convert it to an IF of 100Mhz or so, and follow it up with a direct conversion receiver at 100 Mhz with 20 khz bandwidth.

2) the upconverting local oscillator is a VCO that is controlled

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by a sweep generator. the sweep is controlled by a PWM modulated signal in the audio range.

3) the sweep generator input is connected to the output of a PC sound card. the output of the direct conversion receiver is connected to the input of the PC sound card.

Now, by clever programming of the sound card on the PC, we can make the VCO sweep our passband of interest. The sound is often digitised at 16 bit levels (in the better systems at 32-bit level). This will effectively give us 90db range. the logarithmic scale can be implemented in software. DSP can be used to set the bandwidth to any particular size.

The most important benefit of this design will be that even hams without expensive oscilloscopes will be able to easily make a PC based analyser that is easy to assemble and use.

If there isn't any glaring problem with this design, i would like to pull out my soldering iron and take a go. Is anyone here with spectrum analyser experience willing to share knowledge?

I have recently completed a spectrum analyser, see

http://www.hanssummers.com/electronics/equipment/spectrumanalyser/index.htm.

It is awaiting possible magazine publication so there are not yet any circuit or construction details on the page above. If you want the full details, email me privately and I'll show them to you.

I also tried a direct conversion receiver initially. It doesn't work on in analogue (i.e. non-PC) analyser, because there are all sorts of heterodynes of the sweep frequency against the directly converted incoming signal. Of course I kicked myself afterwards for not thinking of it in advance to save myself the time of the experiment.

I think broadly speaking the final IF should be substantially higher than the frequency of the sweep waveform, so that the final filtering works faster than the sweep. There are lots of people in this forum far more advanced than me who will probably be able to explain it better in terms of filter response times or group delays or something.

Though there might be a way of untangling everything in software so it may work. Being direct conversion you'll also have both sidebands present, which will create further complications. Again, clever software might untangle it but I think it's far from straightforward.

Another problem is the narrow bandwidth. 20KHz is a nice bandwidth to have but I think in a spectrum analyser you also want wider bandwidths available. In particular, if you are digitally generating your sweep voltage, and trying to cover the whole 100MHz, you need of the order of 100,000 / 20 = 5,000 discrete measurement intervals. You can't display that many horizontal pixels on screen. You could average them in software, but at the low 20KHz bandwidth, you're going to need quite a slow sweep rate. 5,000 measurements are a lot and will take a long time.

It's a nice idea but I don't think it will work as it stands.

My recommendation would be to add a 2nd IF to your design, 2nd IF amp and logarithmic detector. In my design I used a 145MHz 1st IF, so the VCO sweeps 145 - 290MHz. The 2nd local oscillator is at 153MHz for an 8MHz 2nd IF, amplified then passed into an AD8307 logarithmic amplifier. Anything similar would work well. I used an SA602 front end for simplicity, but a diode ring mixer would give potentially better performance than the 65-70dB dynamic range I achieved.

You can still use the PC for a nice display, rather than an oscilloscope. Just feed the log output into your PC sound card, and have the PC sound card control the sweep as you suggest. I think you'll solve a lot of problems by adding these few extra modules to the

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analogue front end before introducing the PC. Incidentally, this is exactly what I'm doing with my Mk2 analyser, see

http://www.hanssummers.com/electronics/equipment/spectrumanalyser2/index.htm.

Now I'm hoping for someone to put it all more clearly and professionally than I have here ;-)

I'm not sure that it is necessary considering the base of information on spectrum analyzers of all kinds that have been available for over a half century, nearly all built for military/commercial markets..

22 years ago, Intel had a nice little audio range spectrum analyzer as a laboratory development breadboard-trial thing. Used NO "sweep oscillator" but simply sampled the audio input and did the necessary math calculations to convert from time domain to frequency domain. Quite obviously Intel was promoting their microprocessors in a (then) new niche application. However, that little spectrum analyzer was about as direct-conversion as could be possible. Many, many home entertainment market music systems now build in such spectrum analyzers and show the coarse bar-graphs on front panels in real- time displays.

Given that time now is two decades later with companies such as Analog Devices marketing a number of very fast analog-to-digital converters AND wide dynamic range true logarithmic detectors, a single-conversion spectrum analyzer with a personal computer as the control and read-out can be a viable home project.

To Ashhar Farhan:

..."warming up the soldering iron" and getting going on it is NOT a good thing without a lot of other considerations. First of all, one must have some personal computer program development capability to write the GUI program to show the output display. That is NOT for beginners in programming. Secondly, any sort of DSP program set MUST be done according to DSP math and with known levels of input to a digitizer. Understanding the guts of DSP processing is a prodigious project all by itself.

Third, one has to have some sort of planning ahead of time on what is expected in performance of this spectrum analyzer: Range, band- width, dynamic range, resolution, accuracy. Performance of ANY of those criteria will be affected by the ability to design them into the project AND at what cost. Such things do not assemble like Lego blocks.

Fact: Direct, real-time spectrum analysis exists in audio/music system equipment and has for at least a decade. House-number single IC packages exist to do that analysis and drive simple displays at relatively low cost.

Fact: Digital sampling oscilloscopes today handle 500 MEGA samples per second with resolution exceeding 8 bits per sample. Display of stored digitized information is integral to the oscilloscope, now commonly done on an LCD flat screen. Cost is relatively high.

Fact: DSP techniques, algorithms, ICs exist today to enable frequency conversion, sampling, filtering to be done at relatively moderate to high cost.

Combination of any of the above is possible to make a truly direct spectrum analyzer or single-conversion analyzer with an input range up to VHF. The concept of truly direct spectrum analysis has already been done as a consumer market item accessory. DSOs of today from Tektronix and LeCroy already contain varying degrees of DSP built in. The development of such a spectrum analyzer will be time- consuming and not, in my estimation, an easy project for the casual home hobbyist.

Len Anderson retired (from regular hours) electronic engineer person

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A Morsodist Challenge

From: "Dave Bushong"

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<dbushong@adelphia.net>
Subject: Re: A Morsodist Challenge YKXWBSX7I6 01/03/2004
Date: 2003-10-29 21:23

Hans,

I'm at a loss for words. I bow to the master.

73, Dave

Hans K0HB wrote: [...]

Dear Reverend Dave,

I'm shocked that you haven't heard of one of the great religions of Amateur Radio. Morsodism has been widely discussed here on rrap, but you might be an ignorant newcomer here, so I'll fill you in.

So you want to be a Morsodist. It's not easy, you know: you'll have to impose the stiffest constraints on your mind, accepting authority without question -- you will be subject to jeers and persecutions at the hands of the Spawn of NoCode, and you will suffer during your entire lifetime trying to throw pearls before the swine. But in the end, you will receive your reward from the Almighty in Heaven, and you will have the satisfaction of seeing all others being tossed into the Lake of Fire and QRM.

But enough of that; you're not there yet. You must first prove yourself worthy of being a Morsodist.

Lesson One: Developing Your Psyche

Right now I would like you to relax. That's it -- just sit quietly and let your mind go blank. Concentrate on one thing and one thing only: concentrate on your thumb. Now that thumb is no ordinary run-of-the-mill thumb: what you see is THE THUMB. That Thumb is your world. Nothing besides that Thumb concerns you in the least. As far as you are concerned, that Thumb rules over all. The only thing that should be in your mind right now is your Thumb and its absolute authority over your life. Each and every line, ridge, wrinkle, and bit of cuticle is a Divine Mandate, eternal in its wisdom, power, and relevance. That Thumb was made by MORSE, and its sole purpose is to guide you to your Salvation.

Now, while still concentrating on that Thumb, begin to think about the world around you. Careful! Don't try to take in everything at once: start with the rest of your fingers, and slowly spread from there. See how your fingers are connected with and dominated by your Thumb. Now, slowly extend your perception to embrace your forefinger. It too was created by Morse, and it is all guided by the Thumb. Between your thumb and forefinger lie all the secrets of The Keys of Morse. The rest of the universe is of no concern.

Lesson Two: Establishing Your Doctrine

Congratulations! You've come a long way, but you're not there yet. What I want you to do now is to go to the nearest book store and purchase a copy of the FCCRuleBook. Preferably, it should be bound in nice, shiny black leather, and should have gilt edges and a red silk ribbon. It should be fairly hefty in size, but not so large that you cannot hold it with one hand. Finally, it should have the words of Samuel in red letters, for as you know the Son of Morse only spoke using red letters.

Now that you have yourFCCRuleBook, read it, and read it thoroughly. I imagine that some of it will fit in perfectly with your beliefs and morals, since you have been inspired by Maxim to attempt this indoctrination into Morsodism. As you read, you must remember that your beliefs are the ONLY right beliefs, and all others are falsehoods and are totally incompatible with Morse Divine Truth.

Now you may be dismayed as you read through the FCCRuleBook to find that much of it is outside your beliefs, or may even contradict them. No problem! All you have to do is take a black marker pen and blot out those paragraphs which

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you don't like. Simplicity itself! Now don't you worry if it seems that more of the FCCRuleBook is blacked out than left behind: just remind yourself that the remaining words are emphasized all the more.

Lesson Three: Living a Morsodist Lifestyle

Well, I'm sure that you Novices are simply bursting with questions on how to apply your new-found faith to your lives. I am pleased by your enthusiasm: such zeal is a necessity if you want to begin to work your influence on this filthy and sinful world and mould it into a clean and upright one. Those Spawn of W5YI are simply itching to be preached at, whether they want it or not, so let's get cracking.

First, obtain for thyself some sort of access to the internet newsgroups, specifically any (and all) newsgroups in the "rec.radio.amateur.*" lineage. Never mind that only one of them, the "policy" group is fit for Morsodists like yourself; you must spread the word ALL the heathen NoCodists. Crosspost like crazy, verily even if attacked by Horrible Herman, the Lord High Newsgroup Purity Defender.

As you preach the word of Morse to others, you will inevitably run into those who not only doubt the veracity of your words, but who also pose trick arguments inspired by Lenny Liberal and his henchman, Crazy Charlie, the head Oracalist, in an attempt to trap and discredit you. This lesson deals with some of the major swords wielded by various tempters, and gives you some ideas on how to parry those weapons.

Technobabble scholars: these vermin use their high-falootin' technical jargon and book-learning to pervert the words of the Morse and read things into Part 97 which aren't there. The FCCRuleBook was written by Morse for YOU to use, and contains EVERYTHING you need to know for Salvation, plainly stated in black and white and red. There is nothing magic or mystical about it: it speaks out in the Words of Morse and speaks directly to YOU, as you have personally modified it to your belief set.

If other Spawn of NoCode say that the FCCRuleBook cannot speak the utter Truth because it is inconsistent in what it says, you can reply that there is no inconsistency in your FCCRuleBook: everything that you have in your FCCRuleBook perfectly supports your beliefs (remember Lesson Two). If they quote passages from the FCCRuleBook which they claim contradict your beliefs, you can reply that Oracle has inspired them to pervert the words of Morse and to twist their meaning around in order to try to subvert the Truth. If they say that you are being totally ignorant and irrational in your absolute belief in your Morsodist Doctrine, you can reply that they are being totally ignorant and irrational in their absolute refusal to believe in your Doctrine. And if they say that you are shutting your eyes to reality, you can reply that they are shutting their eyes to Morse.

Peace to you!

Hans, K0HB Lord High Liberator of the Blue Electric Smoke

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Intermodulation Performance and Measurement of Intermodulation Components

(Continued from page 1)

through a device whose output is not a linear function of its input, harmonics of f1 are generated, i.e. 2f1, 3f1, 4f1, 5f1, etc (no device is perfect and so harmonics are always generated even at low levels).

Now, if two separate frequencies exist together in a non-linear device, sum and difference frequencies are also produced in addition to the harmonics. This can be shown mathematically to be the result of a multiplication process between the two original frequencies and hence the two new frequencies are called products. If the two original frequencies are f1, f2 and the highest frequency is f2, then we can expect two other components (or products) of (f1+f2) and (f2-f1). However, it doesn't stop there. Since there are harmonics of f1 and f2, then there will be sum and difference products between all of the harmonics and the fundamentals and between each other. These are the intermodulation products which are frequency components distinct from the harmonic components discussed in the previous paragraph. Of course, if there are more than two fundamental frequencies, then the multitude of products is compounded further.

It can be shown, using a mathematical series, that when harmonics are generated, the harmonies extend upward in frequency to approach infinity, progressively decreasing in amplitude as the frequency increases. Likewise, the intermodulation products could also be considered to be infinite in number. However, we are only really interested in those of practical significance, that is of such a level that they might deteriorate the quality of our signal beyond an acceptable level.

To examine intermodulation products we will consider two frequencies f1 and f2 and some of the orders of intermodulation products. To define the order, we add the harmonic multiplying constants of the two frequencies producing the intermodulation product. For example, (f1+f2) is second order, (2f1-f2) is third order, (3f1-2f) is fifth order, & etc. Let's consider

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Table 1 – Intermodulation Products

1st Order	f1,	f2	100 kHz	101 kHz
2nd Order	f1+f2,	f2-f1	201 kHz	1 kHz
3rd Order	2f1-f2,	2f2-f1	99 kHz	102 kHz
	2f1+f2,	2f2+f1	301 kHz	302 kHz
4th Order	2f2+2f1,	2f2-2f1	402 kHz	2 kHz
5th Order	3f1-2f2,	3f2-2f1	98 kHz	103 kHz
	3f1+2f2,	3f2+2f1	502 kHz	503 kHz
Etc.				

Table 2 – Odd Order Products

3rd Order	2f1-f2,	2f2-f1	99 kHz	102 kHz
5th Order	3f1-2f2,	3f2-2f1	98 kHz	103 kHz
7th Order	4f1-3f2,	4f2-3f1	97 kHz	104 kHz
9th Order	5f1-4f2,	5f2-4f1	96 kHz	105 kHz
Etc.				

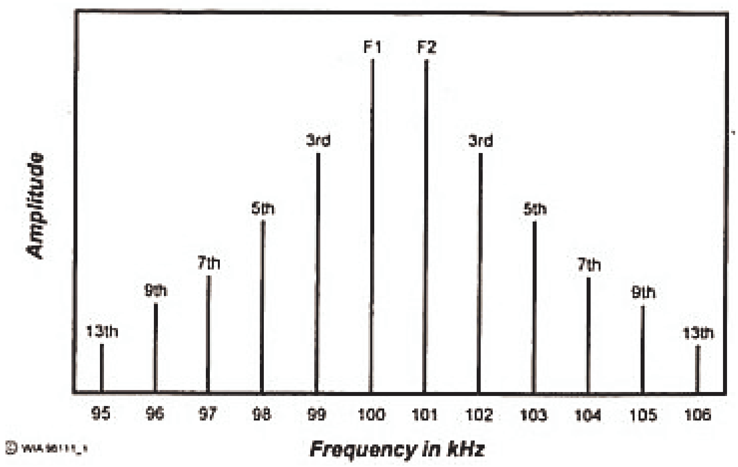

Figure 1 - Spectrum of Intermodulation Components

Intermodulation Performance and Measurement of Intermodulation Components

(Continued from page 9)
f1 and f2 to be two frequencies of 100 kHz and 101 kHz respectively, that is 1 kHz apart. We now prepare Table 1 showing some of the intermodulation products.

Looking carefully at the table, we see that only the odd order intermodulation products are close to the two fundamental frequencies f1 and f2. One third order product (2f1-f2) is 1 kHz lower in frequency than f1 and another (2f2-f1) I) is 1 kHz above f2. One fifth order product (3f1-2f2) is 2 kHz below f1 and another (3f2-2f1) is 2 kHz above f2. In fact it is the odd order products which are closest to the fundamental frequencies f1 and f2.

Let's expand further the odd order products as shown in Table 2.

The series of odd order products can be seen to descend and ascend progressively in increments of 1 kHz from the two fundamental frequencies f1 and f2 respectively. A typical spectrum produced could be depicted as shown in the chart of Figure 1.

Of all the harmonics and intermodulation components produced, are often only interested in those which fall in. the pass band of our equipment and, in the case of the intermodulation components, those which happen to be closest to our fundamental frequencies. The third order components are the closest and also usually the highest in amplitude. Because of this, they are usually the products of most concern and are those which are commonly measured and defined in transmitter and receiver performance specifications.

Effects of intermodulation components

The existence of intermodulation components affects the performance of equipment in various ways. First let's look at audio amplifiers. The presence of any component at the output of an amplifier, but not fed into it, degrades the quality of the signal being amplified. We call this distortion, which can be the result of non-linearity in the amplifier causing the generation of harmonics of the signal frequencies and, in turn, intermodulation components. So we have harmonic distortion and intermodulation distortion which can individually be defined. We often see intermodulation distortion abbreviated to IMD.

Some different effects can be experienced when nonlinearity exists at RF in a transmitter and, in particular, in the final linear amplifier of the transmitter. Consider the amplifier delivering sideband components to the antenna at radio frequencies and because of nonlinearity, harmonics of the various sideband components are generated plus various intermodulation, components. The harmonic components and the even order intermodulation components will be well spaced away from the operating frequency, and hopefully attenuated by the tuned amplifier tank circuit and the antenna tuning system. Not so for the odd-order intermodulation components which are closely spaced around the fundamental components from which they were generated. First of all they will show up as audio distortion after being received and detected by the radio receiver. However, that's not all! We have seen from the previous paragraphs that the odd order components spread out either side of the fundamental components in progression gradually decreasing in amplitude. The effect is to broaden the radiated signal and in receiving the signal, we experience the familiar sideband splatter. As most of us well know, this causes interference to others trying to use another channel near in frequency.

Another application where those odd order, intermodulation components are of considerable concern is in the first Mixer stage of a superheterodyne receiver. The special function of the mixer stage is to produce some
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Intermodulation Performance and Measurement of Intermodulation Components

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form of non-linearity so that an intermediate lower frequency is formed from the sum or difference between the incoming RF signal frequency and a local oscillator frequency. The mixer stage is, therefore, a prime spot for other intermodulation products which we might not want. Let's look at an example. Our receiver is tuned to a signal on 1000 kHz but there are also two strong signals, f1 on 1020 kHz and f2 on 1040 kHz. The nearest of these (f1) is 20 kHz away and our sharp intermediate frequency (IF) stage filter of 2.5 kHz bandwidth is quite capable of rejecting this signal. However, the RF stages before the mixer are not so selective and the two signals f1 and f2 are seen at the mixer input, free to produce intermodulation components at will. Now work out the third order intermodulation component (2f1-f2) and we get (2x 1020-1040) = 1000 kHz, right on our signal frequency. This is just one example of how intermodulation components or out-of-band signals can cause interference within the working band.

Another form of interference in receivers which results from the mixing of intermodulation components is Cross Modulation. This becomes more apparent when dealing with AM signals and the modulation on a strong out-of band signal transfers itself across to modulate the signal being received. The process is probably complex but, due to non linearity in the receiver, one can well imagine the carrier and sideband frequencies of the out-of-band modulated signal mixing to produce difference second-order components at the audio frequencies of modulation. Due to the same non-linearity, the unwanted audio components intermodulate the signal being received. If the receiver is designed for good intermodulation immunity, it will also have good cross modulation immunity.

Receiver IMD Performance

To assess the receiver for its tolerance against interference from internally generated intermodulation products, the receiver is tested for its sensitivity to third-order products using two equal level signals fed to its input and typically 20 kHz apart. The receiver is tuned to the frequency of one of the third order products derived from the two signal frequencies. The level of the combined signals at the input is adjusted until the detected output level is equal to that generated by the receiver's self noise. That is, there is a 3 dB change in output level between when the signals are on and when they are off. The level of the third order product necessary to produce output equal to the receiver "noise floor is recorded. For the purposes of following discussion we will call this level IMD Threshold.

We also need to know the normal signal input level which produces audio output equal to that of the receiver's own inherent noise or its noise floor. This is done by tuning the receiver: to one of the two frequencies and again adjusting input level to give an output level 3 dB above the noise output level. For the purposes of the discussion we will record this input level as the Signal Threshold. The difference in dB between the IMD threshold and the signal threshold is called the Intermodulation Dynamic Range or IMD Dynamic Range. The higher the difference, the better the immunity to interference from IMD products.

Now, there is an important characteristic of the third-order products which makes their presence more of a problem than one might first imagine. Assuming no compression (due to AGC, etc), output from a fundamental signal is proportional to input, ie for 10 dB rise in input level there is 10 dB rise in output level. However, the output of the third order product is proportional to the cube of signal input level and, for a 10 dB change in input level, the product increases by 30 dB.

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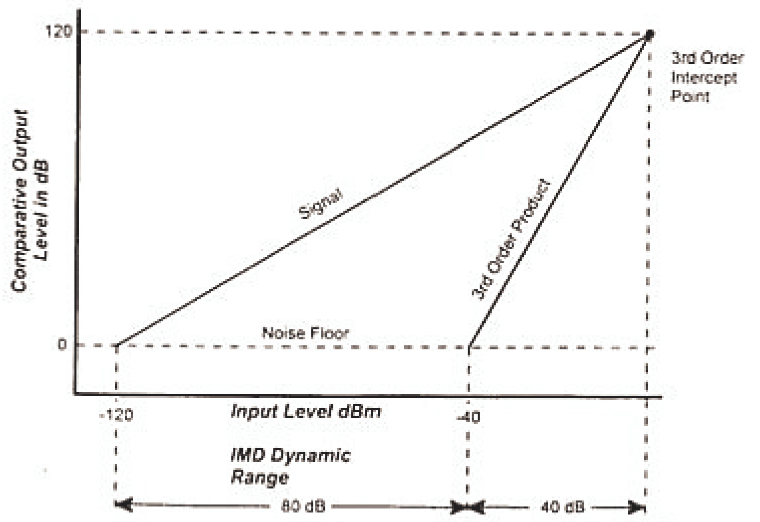

Figure 2 - Receiver Intermodulation Performance Curves

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Now refer to Figure 2. One curve plots a linear rise in output level against input level for the fundamental signal frequency. The other plots the level of third-order IMD products against input level, the output rising 30 dB for every 10 dB change in input level. At an output level equal to the noise floor, the two curves are separated by an input level difference equal to the IMD Dynamic Range. As the intermodulation curve rises with greater slope than the fundamental curve, they cross at a point called the Third-Order Intercept Point where the intermodulation product output level is equal to the fundamental signal output level.

The Third-Order Intercept point is normally a theoretical point well above the receiver overload level. However, it is often specified to define intermodulation levels, and particularly in specifications for mixer packages.

The Third-Order Intercept point can be derived on decibel scales by first extending the signal curve linearly from the signal threshold point on the noise floor axis so that output level increase in dB equals input level increase in dB. Mark the IMD threshold point on the noise floor axis. This is at an input level higher than signal threshold by an amount equal to the IMD dynamic range. Mark another point on the noise floor axis beyond the IMD threshold point by an amount equal to half the IMD dynamic range. Extend this point vertically to cross the signal curve and this is the Third-Order Intercept point. Join this point to the IMD threshold point to complete the curves such as is shown in Figure 2. In the diagram, the noise floor input level is -120 dBm, the third order components become detectable at 40 dBm and the IMD dynamic range 80 dB. The theoretical third-order intercept occurs at an input level of 120 dB above the noise floor input level.

It can be seen from the curves of Figure 2 that, above the IMD threshold level, the IMD products can become quite a problem. In the example, IMD products from out-of-band

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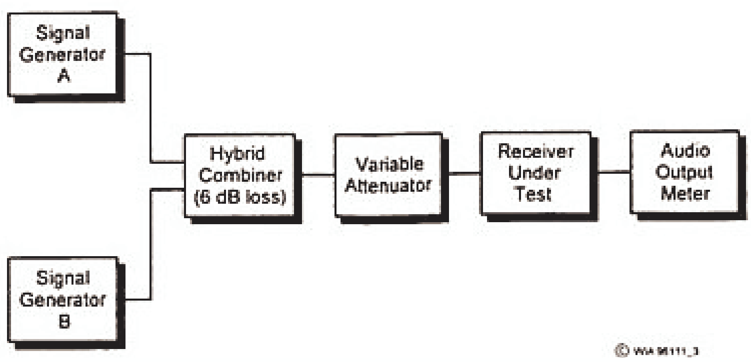

Figure 3 - Testing of Receiver for Third -order Intermodulation Performance

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signals at an input level of -40 dBm would barely be apparent. Increase the level by a mere 10 dB and the interference from these products would increase by 30 dB.

Receiver Measuring Gear

To carry out intermodulation performance on a receiver, the set-up shown in Figure 3 is required. The RF outputs of two signal generators are combined in a hybrid circuit designed to prevent interaction between the two generators. A hybrid circuit is balanced so that a signal at any one input port cannot reach the other. However, both signals appear combined at an output port.

The combined output is fed to the receiver via an adjustable attenuator with a range up to around 80 dB and resolution of 1 dB. Assuming that the receiver has an input resistance of 50 ohms, both the combiner and attenuator are designed for a circuit impedance of 50 ohms.

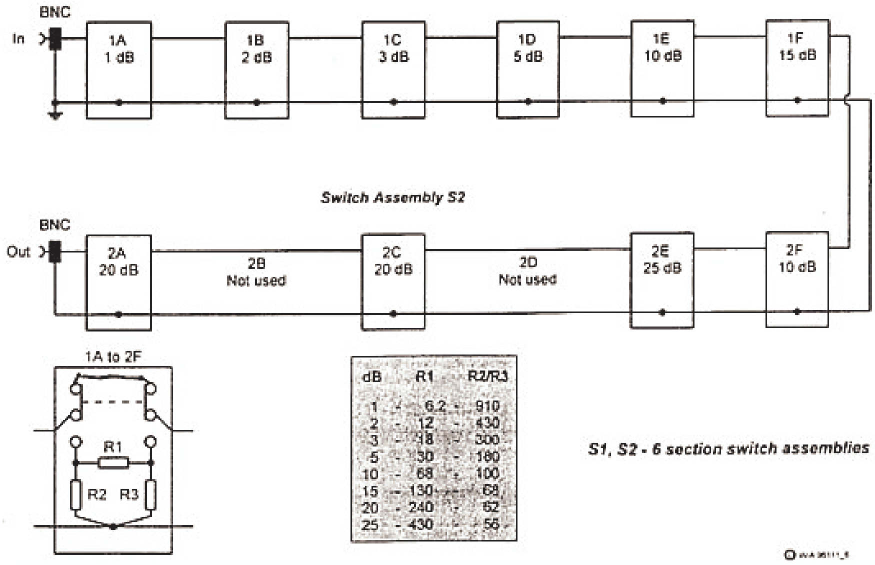

Figure 6 -Simplified Circuit Diagram of the 50 ohm Attenuator

A hybrid combiner and attenuator assembled by the writer is shown in Figure 4. The hybrid circuit, Figure 5, is one taken from the ARRL Handbook. The combiner has an insertion loss of 6 dB for each signal channel.

The attenuator was made up using two mechanically interlocking, in-line switch assemblies of the type similar to those used in older style push-button car radios. The assemblies, each of six switches, were recovered from some old intercom units and each switch came with plenty of change/over contacts to switch in or out an attenuation pad. One assembly switches in 1 dB, 2 dB, 3 dB, 5 dB, 10 dB and 15 dB pads. The other switches in 10 dB, two of 20 dB and 25 dB pads. Up to three switches on each assembly can be simultaneously pressed to lock in so that up to six pads can be in circuit together to provide a continuous selection of total attenuation between 1 and 95 dB. The circuit diagram of the complete attenuator is shown in Figure 6.

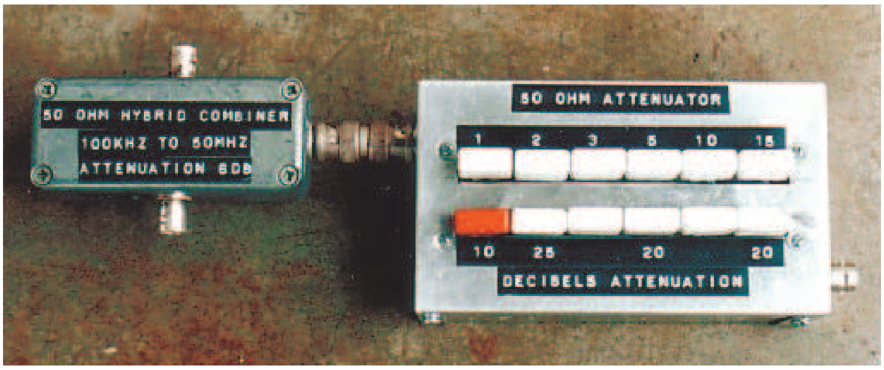

Figure 4 - Hybrid Combiner and 50 ohm Attenuator assembled by VK5BR

50 Ohm Attenuator

The only other device neces-

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sary is some form of AC voltmeter to measure the comparative level of audio signal at the receiver output. All it is required to do is to record a 3 dB change in level above the receiver noise floor. In terms of voltage increase, this is a rise of 1.4 times.

Our references have so far been made to levels in dBm, or decibels referred to one milliwatt. However, signal generator outputs are commonly calibrated in microvolts and millivolts with scales in multiples of 10. To convert between units, 1uV across 50 ohms is -107 dBm. Each time the voltage is multiplied by 10, add 20 dB so that 10uv is 87 dBm, 100 uV is -67 dBm, etc.

To find the signal threshold, set one signal generator to a fairly low level (say 10uV or -87 dBm) and tune the receiver to the signal generator frequency. Adjust the attenuator so that the signal raises the audio output signal just 3 dB (1,4 times volts) above the noise level (measured with signal off). The signal threshold in dBm is equal to -87 dBm, minus the loss in dB set by the attenuator, minus 6 dB loss in the hybrid combiner.

To find the third-order IMD threshold, set the two generators (20 kHz apart in frequency) to an equal level somewhat higher, such as 10 mV (-27 dBm). Tune the receiver to the frequency of one of the third-order products. Adjust the attenuator until the audio output from the third-order signal is just 3 dB above the noise level. The IMD threshold is equal to -27 dBm, minus the loss in dB set by the attenuator, minus 6 dB loss in the hybrid combiner.

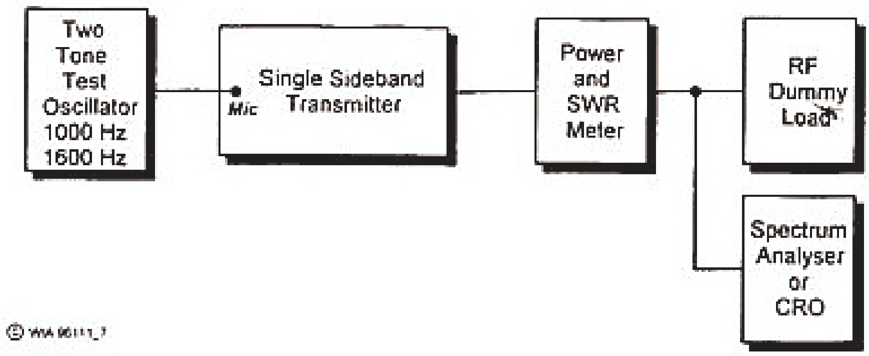

Figure 7 - SSB transmitter test arrangement for IMD performance

Transmitter Tests

To check out a single-sideband transmitter for those intermodulation components which cause sideband splatter, we need a two tone audio generator to feed into the microphone input of the transmitter. This can be quite a simple test unit, consisting of two fixed frequency oscillators delivering the same output level into a resistive network which combines the two signals. Simple two tone generators have been presented in Amateur Radio from time to time. In the March 1983 issue, the writer

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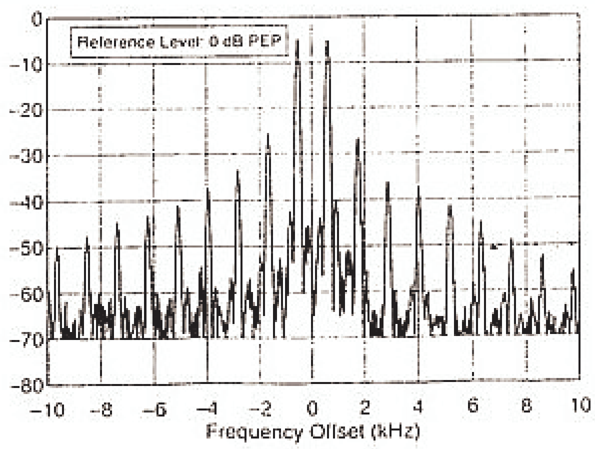

Figure 8 - Typical Spectral display from the RF output of a SSB transmitter using two tone modulation and showing the intermodulation products generated (sample from ARRL test in March 1996 issue of QST).

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described one using two FX205 Tone Generator packages. In this one, frequencies were set at 1000 Hz and 1600 Hz.

A test arrangement for the transmitter is shown in Figure 7. The two tone oscillator level is adjusted to provide full RF power from the transmitter into a dummy load. Power can be monitored with the usual Power/SWR meter. Apply the audio signal in short bursts as most single-sideband transmitters are designed for speech and the final amplifier stage might be damaged if sustained on continuous full power. The best way to monitor the level of the various intermodulation sideband components is to examine the RF output signal using a Spectrum Analyser.

Figure 8, taken from March 1966 issue of QST, is a typical spectrum analyser display of the RF output of a single-sideband transmitter fed with two audio tones 1000 Hz apart. Two fundamental RF sideband frequencies are created but we can also see a family of odd-order intermodulation frequencies, either side of the two fundamentals with all frequencies spaced 1000 Hz apart. The display shows that the third order products are around 21 dB below the fundamentals, the fifth order 30 dB below, the seventh order 33 dB below, & etc., in decreasing amplitude as the order progresses.

Whilst the spectrum analyser is the order of the day in the modern electronics laboratory, not many radio amateurs could boast of one in the radio shack. However, the Cathode Ray Oscilloscope (CRO) is a more common piece of test gear, and with this we can get some idea of whether there might be an excessive spread of intermodulation sideband components. Figure 9, taken from the ARRL Handbook, shows CRO displays of the RF output generated from a two tone audio source fed to the transmitter. In diagram A, the waveform is quite good and we could expect a fairly clean signal transmitted. In diagram B, compression of the waveform peaks is occurring, possibly because the final amplifier is being driven too hard into a state of poor linearity. If there is poor linearity, then we can expect intermodulation components to be generated and sideband splatter.

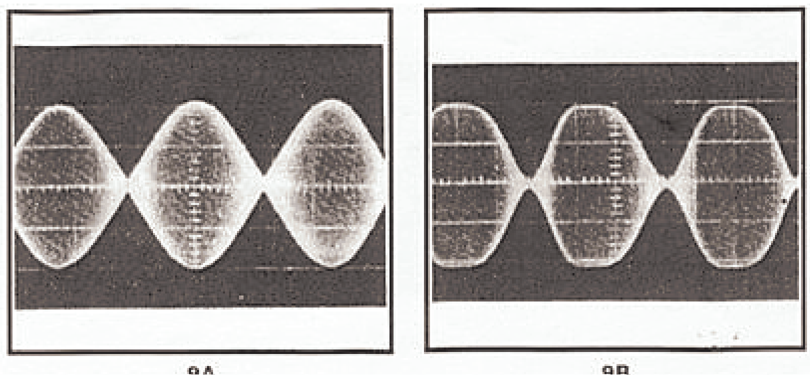

Figure 9 - SSB Two Tone test showing RF waveform on a CRO (a) Good waveform (b) Peaks compressed (IMD high, sideband splatter).

Another test that might be applied is to demodulate the transmitter so that we get the two tones back as audio. Perhaps a station receiver can be used for this purpose if it can be prevented from being overloaded by the transmitter. The audio intermodulation distortion tests as described in the following paragraphs can then be carried out. The tests could apply to any mode of transmitter and matching receiver whether it be SSB, AM, or FM (for AM, a simple rectifier and RF filter would be adequate for demodulation). The only problem with this form of test is that the distortion measured is the combined distortion of both transmitter and receiver. If excessive distortion occurred, one would have to be certain that it wasn't caused by the receiver.

Measurement of Intermodulation Distortion at Audio Frequencies

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As with other tests described, two audio signals at different frequencies are fed through the device to be tested and the output is monitored. If a modern spectrum analyser is available, the relative amplitudes in decibels from all components at the output can be displayed. As the X axis of the analyser display is calibrated in frequency, the various intermodulation components can be identified and their amplitudes recorded relative to the two fundamental frequencies.

Another instrument of a past era, but which can do a similar job, is the Heterodyne Wave Analyser. This is, in effect, a sharp tuneable filter which achieves its sharpness and tuneability by heterodyning the measured signal with a tuneable oscillator and passing the difference frequency through a sharp fixed 50 kHz crystal filter. By adjusting the tuneable oscillator, the various frequency components can be selectively tuned in and the outputs at 50 kHz can be compared. The Wave Analyser was described in the writer's previous article on Measurement of Distortion, Amateur Radio June 1989.

Another method to measure the intermodulation level is to make use of a CRO display as shown in Figure 10. Two audio signals of widely different frequency are combined and fed into the device under test. The lower frequency signal has an amplitude four times that of the higher frequency signal. The output of the device is fed to the CRO vertical plates via a high pass filter which removes the low frequency signal. The CRO time base is externally synchronised to the low frequency signal. Intermodulation is shown on the display as an amplitude modulation waveform of the lower frequency on the higher frequency carrier. The reason for the four to one signal amplitude ratio is to amplify the apparent modulation and improve resolution in reading the display. The test set-up, shown in Figure 10, uses a 100 Hz low frequency signal and a 2000 Hz high frequency signal. A simple resistive mixing network is used to prevent interaction between the audio generators. Referring to Figure 11, percentage intermodulation is calculated from a and b scaled on the CRO display as: % Intermod = (a-b)/2 (a+b) x 100.

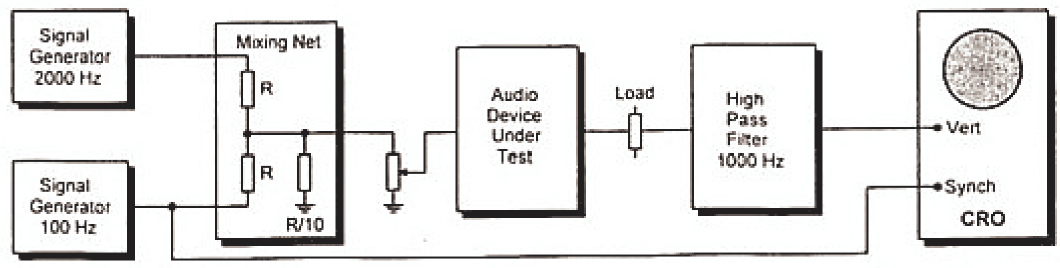

Figure 10 - Test for audio intermodulation distortion using CRO display.

In this test, it should be clear that we are essentially measuring the effect of the second-order intermodulation components at (f2-f1) 1900 kHz and (f2+f1) 2100 kHz. This should not be confused with the fact that at radio frequencies we had been mainly concerned with the odd-order products because it was those which appeared in close to our tuned band. However, at audio frequencies, both odd and even products fall within the audio band.

Summary

Intermodulation products have been discussed with particular attention to how their presence affects our transmitter and re-

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Intermodulation Performance and Measurement of Intermodulation Components

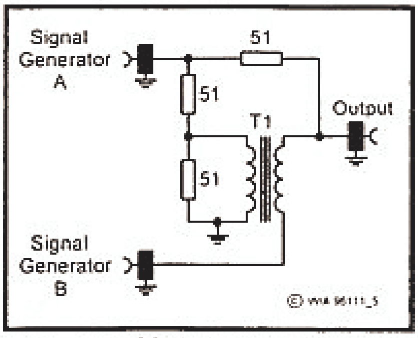

Fig 5 - Hybrid combiner (based on a circuit from the ARRL Handbook, ref 2). T1 consists of 10 turns bifilar wound on FT50-72 core.

(Continued from page 16)
ceiver circuitry. In audio circuits, they are one of the contributing distortion factors which deteriorate audio reproduction quality. At radio frequencies in transmitters, they appear as what we recognise as sideband splatter. In receivers, circuits susceptible to intermodulation encourage interference from signals outside the receiver pass-band.

Various ways have been explained as to how intermodulation components can be measured and how the equipment performance in terms of IMD susceptibility can be specified. The reason why the third order performance is usually defined in RF circuits has also been discussed.

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Taken from:

HomePage <http://users.tpg.com.au/users/ldbutler/index.htm>

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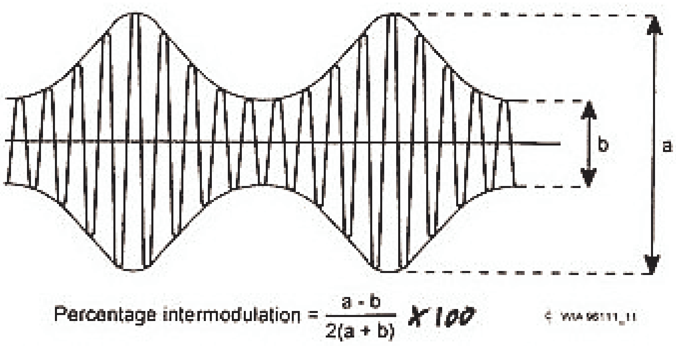

Figure 11 - CRO display waveform of intermodulation and method of measurement

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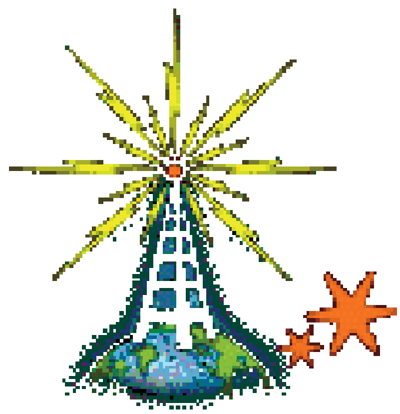

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