A PRACTICAL EVALUATION AND COMPARISON OF SOME MODERN DATA MODES

as applied to

THEIR POTENTIAL FOR UK-WIDE AMATEUR EMERGENCY DATA BROADCASTS

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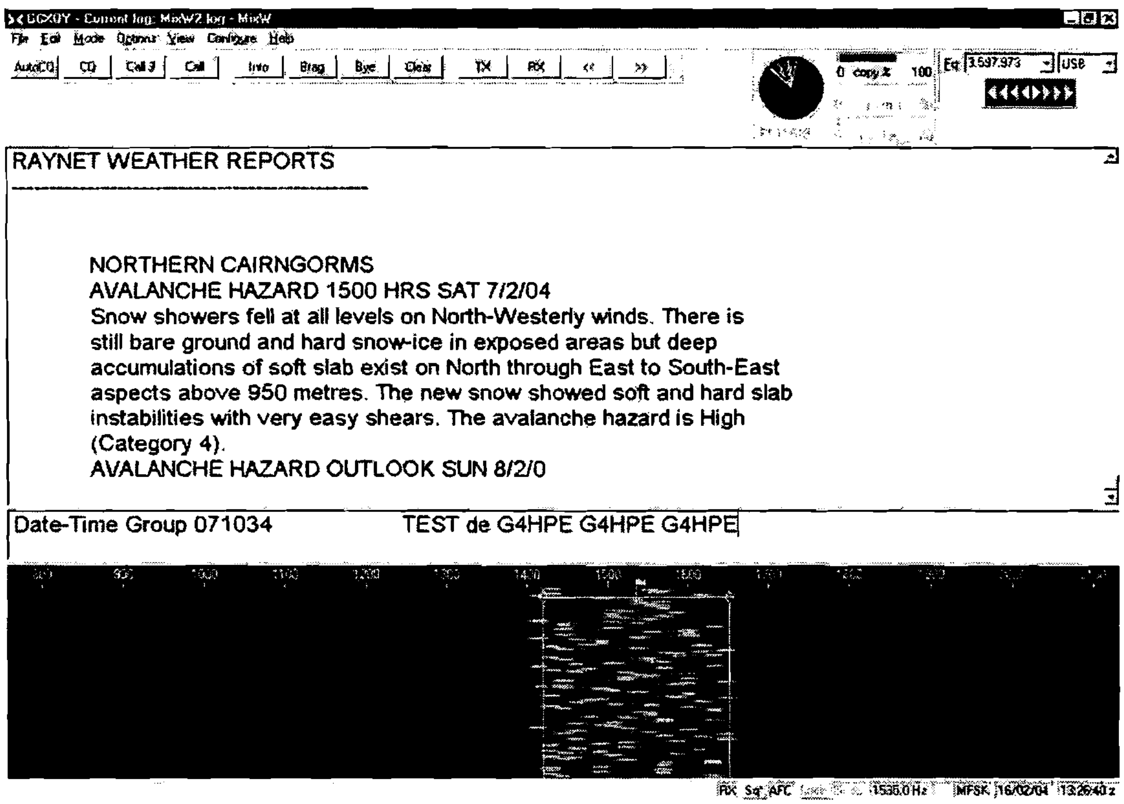

INTRODUCTION

This report aims to make recommendations for the use of modern data modes when considered for the purpose of broadcasting textual information (few transmitters to many receivers) over Amateur Radio HF paths.

In recent years, the advent of *Digital Signal Processing* (DSP) using the soundcard of a standard home computer has enabled great advances in the control of the sound spectrum. Principally, the techniques of audio signal manipulation and filtering have moved from external mechanical devices to software control directly within the computer.

Such advances have led to the development of several new data transmission modes that perform well over HF radio paths. Much of the software is freely available and simple to operate by the Radio Amateur, using modest equipment at home or in the field [see **Figure 1**].

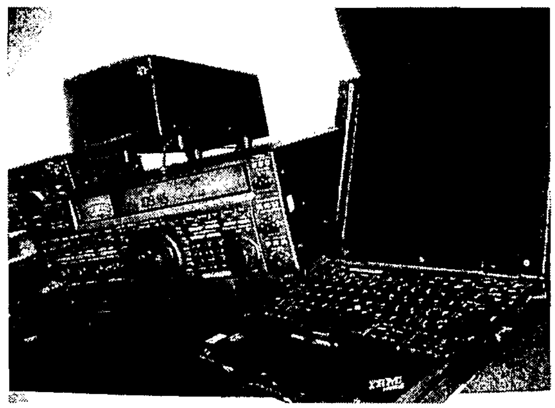

Figure 1: The author's data station

The picture shows the laptop running MIXW2, interfaced with the Icom IC775DSP.

The advantages of the newer data modes over the older include: greater speed and accuracy for a given spectral bandwidth, improved robustness against all types of interference, a more compact set-up and much better marginal signal performance.

It is suggested that these modern data modes could find an application in amateur emergency HF communications. The possibility of 'broadcasting' messages and information is of interest here. For example, it may be possible to provide a UK-wide service from a single, or a few, well-appointed stations. Because of the comparative ease with which an HF data *receive-only* facility can be established at temporary locations, a national information system might be an achievable goal for radio amateurs. Such a facility might, for example, carry low-precedence traffic in order to free official channels for more important messages.

THE EXPERIMENTATION

Throughout 2002 and 2003, groups of radio amateurs engaged in a series of on-air trials. The purpose was to examine, in practical rather than theoretical terms, the relative performance of the newer data modes.

The modes examined were [see **Figure 2**]:

- **PSK31**

  Narrow modes featuring one carrier with 31.25Hz sidebands. The idle state is a characteristic pair of carriers spaced 31.25Hz apart. BPSK31 determines 1's and 0's by reversing the carrier phase to a predetermined pattern.

- **MT63**

  Modes that employ 64 separate tones over an audio bandwidth of 500Hz, 1kHz or 2kHz. Forward Error Correction combines with interleaving systems.

- **MFSK16**

  A mode that uses sixteen tones within an audio bandwidth of 316Hz. Convolutional Forward Error Correction is used.

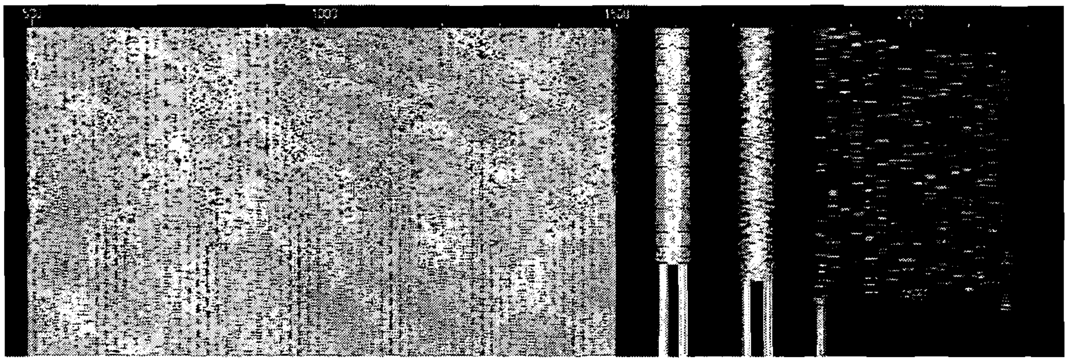

Figure 2: Data modes viewed on the MIXW2 waterfall display

On the left can be seen the large 64-tone block waveform of MT63-1K, then BPSK31, the zigzag pattern of QPSK31 and finally the sixteen tones of MFSK16. The waterfall display shows frequency across the X-axis, and time moving downwards on the Y-axis; the most recently received signal is at the top of the display. Each mode can be seen idling at the bottom of the display just prior to the start of a message.

For the purposes of a practical messaging system, certain other modes were dismissed as having a data rate that was too slow to pass realistic volumes of data within a reasonable time scale. However, it is to be noted that these modes often use minimal bandwidth and can perform very well over marginal paths. Thus, their use should not be dismissed should the importance of the messages outweigh the long time it would take to send them.

Note too that, in the *broadcast* application proposed here, the only modes being considered are necessarily of the 'unconnected' type; in other words, those that do not require 'hand-shaking' acknowledgements from the receiving end. The advantages of not having to transmit acknowledgement packets from the receiving site are several. The means to generate an adequate return signal (a matched, efficient antenna), the power source required to achieve this and automatic PTT switching connections to the radio, are all obviated. However, there are several very efficient new data modes that are of the 'connected' variety (known generally as the 'ARQ' modes) that could be of value in point-to-point links over good paths. The use of these modes at HF is for investigation as a separate study.

Initially, much of the work was carried out under a temporary Notice of Variation to the UK amateur licence using spot frequencies in the 5MHz range. Long text messages were sent via the radio link. The same messages were also distributed by email for precise comparison purposes. As the experimentation progressed, informal transmissions of the RSGB's 'GB2RS News' scripts provided large amounts of source text of topical interest, and a wider group of amateurs were able to report. Data transmissions of GB2RS were also made on 1.9MHz.

Critical to the effective analysis of the test material was an automated method of comparison between the original and remotely received texts. Microsoft Word has a useful 'Compare Documents' facility [see **Figure 3**] and other software, such as 'ExamDiff', were also employed to provide statistical evidence of performance.

TEST RESULTS 17/01/03
Key:
Crossed out in green = not received
Underlined in red = as received

ORIGINAL TEXT (MESSAGE SHORT01)

Elm wood burns like churchyard mould -
E'en the very flames are cold.
Poplar gives a bitter smoke,
Fills your eyes and makes you choke.
Apple wood will scent your room,
With incense-like perfume.
But Ash wet or Ash dry,
For a Queen to warm her slippers by.

AS RECEIVED AT G3LNM (TRANSMISSION 1/3)

Elm wood burns like churchyard mould -
E'en the very flames are cold.
~~Poplar~~etoplar gives a bitter smoke,
Fills your eyes and makes you choke.
Apple wood will scent your room,
With incense-like perfume.
But Ash wet or Ash dry,
For a Queen to warm her slippers by.

Characters totally missed = 0
Extra characters inserted = 1
Characters received incorrectly = 1

AS RECEIVED AT G3ZSU (TRANSMISSION 2/3)

Elm wood burns like churchyard mould -
E'en the very flames are cold.
Poplar gives a bitter smoke,
Fills your eyes and makeseytAakes you choke.
Apple ~~wood will~~woos wil scent your room,
With incense-like perfume.
But Ash wet or Ash dry,
For a Queen to warm her slippers by.

Characters totally missed = 7
Extra characters inserted = 0
Characters received incorrectly = 3

Figure 3: Typical on-air test results, compared using Microsoft Word

When comparing the data modes, it was important to try to eliminate as many variables as possible. If different modes were to be compared over a single path between two stations, power levels and choice of antenna remained the same for both modes and the comparative tests were conducted as closely together in time as possible. In order to provide confidence in results, the same comparisons were conducted many times.

A useful development, conceived by Lionel Sear G3PPT, was the 'COMBO' idea. This combined several data modes into a single waveform, slightly separated in frequency so that they could each occupy a unique position in the same spectrum. The advantage of this system was that the signals were transmitted simultaneously as a group, meaning that exactly the same propagation effects would apply to all of them. By capturing the audio at the receiving end of the link, each data mode could be examined in turn by playing back the resulting sound file into the data decoding software.

BRIEF NOTES ON RELATED ISSUES

The physical implementation of an Amateur Service data broadcast system is outside the scope of this report. However, the following notes may serve to assist those pioneers who aim to design such a resource.

Guaranteeing the radio path

To provide a truly effective broadcast facility, it would be desirable to strive for twenty-four hour coverage of the United Kingdom. While choice of mode might affect this to a degree (as discussed below), the utilisation of appropriate radio frequencies would be the major factor.

The underlying principal would be that, over the course of a whole day, it would probably be necessary to take account of the *Critical Frequency* of the reflective F2-layer (known as 'f_oF2'). Approximately speaking, HF radio waves are only reflected back to earth near-vertically when a frequency below f_oF2 is used. Near-Vertical Incidence Sky-wave (NVIS) communication [1] is particularly effective for medium-range communications in and out of a disaster zone via HF, particularly if the terrain is mountainous, but relies of careful interpretation of f_oF2. The F2 Critical Frequency changes throughout the day, rising after dawn and falling again at dusk. Depending on many contributory factors, the maximum f_oF2 might be as high as 14MHz and the minimum below 2MHz. Considering the use of the lower amateur bands at 1.8MHz, 3.5MHz, 5.4MHz and 7.0MHz, this implies a requirement to change frequency from time to time, or to facilitate multiple transmissions of the same material on more than one frequency simultaneously.

Terrestrial and solar cycles, together with seasonal variations, would also need to be accommodated. It is worth noting that, in general, data modes will continue to perform satisfactorily over poor radio paths long after plain speech communication has been rendered unworkable, meaning that an agreed frequency schedule can be made less volatile.

[1] Near Vertical Incidence Sky-wave Communication, David Fiedler and Edward Farmer, WorldRadio Books, PO Box 189490, Sacramento, CA 95818

Choosing the most suitable data mode

To maximise success, the selection of the most suitable data mode to use would depend on the prevailing condition of the radio path. Many factors affect the various modes in different ways. These might include: simple signal strength, fading, atmospheric and/or man-made noise, multi-path effects including delays, propagation disturbance, static bursts and so on. The stability of the radio installation, particularly considering a temporarily established emergency station, should also be considered.

It is unlikely that one particular data mode will be appropriate for use under all conditions encountered on HF amateur bands. If communications are marginal, it may be necessary to move to a mode that is optimised to function under such conditions even if this is at the expense of speed. Alternatively, should it become necessary to use heavily populated Amateur Bands then a reduction in spectral bandwidth may be called for: again, a change to the most effective data mode would be apposite.

Logistics

The logistic issues surrounding how a national data broadcast system could be realised also fall outside this report. Some of the factors that should be considered are:

- The means by which original data arrives at the transmitter location.
- The format of the material transmitted, from the commissioning authority's point of view.
- How the data would be sorted at the receiving location, and how it would be sent onwards to the recipients locally.
- The means by which repeats of missed information could be requested.
- Practical matters at both ends of the link, such as the resilience of power supplies and the physical interfacing of computers, receivers and antennas.
- The need for a precedence structure to prioritise message handling.

Visual images

The transmission of visual images has not been considered. However, there are a large number of modes to choose from and some can build a good black-and-white image in just a few seconds using audio tones across bandwidths of less than 1kHz. The received files are also in a ready form for manipulation (for example ***.jpg**, ***.bmp** or similar).

The critical computer-to-radio connections

To be truly successful with data modes, a good deal of attention must be paid to the interconnection between the radio equipment and the computer soundcard. It cannot be stressed strongly enough how important it is to maintain linearity, particularly when transmitting. Many experimenters fall in to the trap of compromising the audio interface in their enthusiasm to get on the air and overdriven transmitters are a common occurrence. Not only are the chances of individual success highly diminished, but a splattering transmission is also an antisocial practice that can detrimentally affect other spectrum users.

It is vital that distortion is prevented at every stage throughout the signal path and this will take time and effort to get right. In practice, this means that audio signal levels must be set up both within the computer, the transmitter and receiver. The radio equipment settings for data operation are likely to be quite different from those used for speech. A general guide is given in **Appendix A**. There are two very good articles in RadCom that go into detail about the need for linearity when using PSK31 [2].

There is no doubt that the calibration of the transmitting chain is far more critical than the receiving chain. This is another advantage of using broadcast-type modes where the stations out in the field, already potentially challenged by being located in disadvantageous conditions, can afford to be set up less precisely through the need to be capable of receiving only. If it is necessary to broadcast data from within the disaster zone, careful attention must be paid to the quality of the station build and the linearity of the transmitted signal.

[2] "In Practice", RadCom November 2003 p.94 and RadCom January 2004 p.59

PARAMETERS USED TO COMPARE DATA MODES

Part of the consideration of the most suitable data transmission system must be to decide what value is placed on the various merits of each mode. Factors that are less important in day-by-day amateur contacts may become more significant in the field of emergency communications.

In approximate order of importance *in this application*, the criteria against which the data modes were judged are:

1. Accuracy (error correction)
2. Data rate
3. Weak signal performance
4. Tolerance to interference
5. Efficiency (RF bandwidth and Duty Cycle)
6. Ease of operation
7. Compatibility and Adaptability
8. Reliability
9. Availability

The numbering of these sections is retained through to the results section that follows.

1. Accuracy (error correction)

Has the original data been transferred across the transport medium without modification?

In emergency communications, it is clearly very important to maintain a high degree of accuracy. Consider a message containing details of the quantity of a drug to be administered to a casualty. If the text is even slightly garbled, the recipient could be misled, leading to a potentially life-threatening error. Correct representation of the amount of the drug is critical. The difference between '100mL' and '10mL' is merely the loss or addition of a single character but could have devastating consequences if wrongly acted upon.

It is very important to keep in mind that 'unconnected' broadcast-type modes get no response from the destination. Thus, the transmitting source has no electronic feedback as to whether the data has been received in whole, in part, or not at all. This could be considered as a significant drawback. However, many modes use techniques of *Forward Error Correction* to offset this problem. Powerful 'number-crunching' software adds extra information during encoding that allows the receiving decoder to detect, and usually correct, erroneous parts of the bitstream. Another technique is *interleaving* where the data is sent more than once in the time and/or frequency domain, giving the receiver several chances to capture the information correctly. This technique spreads and muddles up the bitstream before sending (but to a known rule!) so that adjacent sections of the data do not get transmitted in order. When the correct sequence of bits is reconstructed at the receiving end, any bursts of interference therefore do not affect successive parts of the data. Combinations of these devices can lead to quite remarkable degrees of accuracy.

It must be accepted that *total accuracy* cannot be guaranteed when using broadcast-type modes. Luckily, however, the modes with multiple, comprehensive error correction systems tend to mute or put spaces into the character positions where reception is in doubt, rather than fill the screen with random hash. This is an important advantage to interpreting doubtful text. However, achieving such high levels of correction may increase the amount of material it is necessary to send due to the extra crosschecking information that must be embedded. So, there may be a compromise between accuracy and speed; the ratio between the two would be a matter for careful consideration with due deference to the nature of the information being passed.

Of course, the best and final arbiter of accuracy is human intelligence itself. Thus, returning to the medical example above, a knowledgeable doctor would immediately question the quantity of the drug, through experience, if the dose seemed doubtful. A repeat of the message could then be requested (bearing in mind that a robustly designed regime would probably specify that the same message would be transmitted several times over a suitable period in any case).

As an interesting aside, the problem of verifying a vital fact (the 'quantity' in this example) can also be greatly eased by intelligent composition on behalf of the sender. The use of repeated text and the spelling out, in long-hand, of important details can help to reconstruct mildly corrupted text. Using the medical example again and deliberately introducing some erroneous characters:

*'...to administer 10**L (REPEAT one hundr*d millili*res)...'*

It can be seen that the reemphasis of the critical information has probably recovered sufficient meaning to proceed.

Another concerning issue is that of character transformation. Some of the older data modes use a restricted ASCII character set. For example, the '@' (meaning 'at') character was only rarely used in Standard English until the explosive arrival of the Internet and the, now ubiquitous, email address format. Modes that have a limited character set, such as RadioTeletype ('RTTY'), omit this symbol and, perhaps worryingly, ignore it. Restricted character sets can be circumvented to an extent by agreement to use other combinations of characters to represent '@', but this is hardly useful for a modern facility. Although the key numeric and alphabetic symbols are, of course, retained, the way some modes substitute the less common symbols with others is somewhat random and could theoretically change the meaning of the text. For example, unrecognised characters are sometimes replaced by a question mark: clearly introducing the potential for confusion. It should also be noted that some of the more basic modes only allow capital letters.

Perhaps more critical to accurate communications is the preservation of text format, together with pagination and tablature. In this we find a significant shortcoming with much of the software currently available to the Radio Amateur. Although some programs allow the operator to select a font for the visual display on the computer screen, this is purely for local convenience and is not preserved across the data link. Generally, the format of the text to be sent can be likened to the rather limited parameters of the basic text file: there is no choice of font, bold or underlined print. Colour is definitely out of the question! Indeed, in terms of importing text into programs, the only permitted type is usually the ***.txt*** type of file.

Interestingly, email via the Internet can suffer a similar problem: a well laid out HTML document from the sender is often translated into the more basic ***.txt*** format, depending on the settings of the recipient's inbox.

Clearly, such a restriction can diminish the appearance of a well-presented document. Some software will allow direct importation of higher-level file types, such as Word documents, but the process usually involves a rather blunt translation into simple text format. While a good attempt is made to extract the text from the unprintable control characters this is nevertheless a risky strategy if accuracy is to be maintained. So, if a change of file type is necessary, then this is best done from within the originating program that will probably have a designed solution for the purpose.

A further, potentially damaging, effect is the weak perpetuation of line length. Word wrapping at the receiving end is usually an uncontrollable function of the font size and available screen width. It is fair to say that the majority of modes will faithfully carry over an intentional *carriage-return-line-feed* instruction, but otherwise the text on the sending screen is not likely to appear with the same line length at the receiving end.

Furthermore, arranging to print the text onto paper from the computer is likely to be 'as displayed' at both ends. This is a particular problem should the original text contain a *pseudo* table, that is to say one whose columns of data have been lined up by careful use of spaces between them. Firstly, the font in use on the receiving screen may allocate a different width to the space character. Secondly, the columns may automatically wrap if the table is too wide to fit on the receiving screen and the tablature will be completely confused, making the information hard to understand. Setting the same font style and size at both ends of the link is likely to be a good starting point but is still not a guarantee of accuracy.

A good method of preserving the integrity of a document, spreadsheet or other proprietary file is to send it directly, a bit like an email attachment, in binary format. If this facility is available – and sadly this is currently limited to just a few programs - it is theoretically possible to transfer material without modification. However, being packed with additional control characters, such files are usually quite large in comparison with their basic text counterparts and therefore demand a very heavy increase in the amount of data being sent across the link. As an example, a typical Word document (***.doc**) is about fifteen times larger than its equivalent Text file (***.txt**)!

A single page of a Word document, which is approximately 30 kilobytes in size, takes about half an hour to send across a perfect link using MT63 at 2kHz bandwidth. Therefore, a careful judgement as to the importance of the preservation of format is required. That said, there is no doubt that organisations who might use the system are likely to have IT-based message preparation, quite possibly in a format that is unique to their operation which they will expect to sustain, and therefore the resolution of this issue continues to be a challenge. This takes us neatly on to the next criterion.

2. Data Rate

How quickly is the information sent across the radio path?

The speed with which data is sent over a link is generally governed by two factors.

Mainly, the data rate is dictated by the available bandwidth – in layman's terms, the 'fatter' the pipe the more 'stuff' you can pour down it per second. In today's rapidly developing world of communication by cable, bandwidth is becoming irrelevant with high-volume circuits now cheaply available to industry and the home user alike. Transmitting data over radio circuits, however, cannot hope to match cabled bandwidths. At HF, amateur radio links must occupy as little spectrum as possible.

Secondly, the speed of the transfer is affected by the amount of extra data that must be sent to support error correction systems. For example, the message may actually be sent more than once and may be surrounded by 'check-sum' information intended to allow the receiving software to detect and correct errors. Thus, the apparent data rate is slower but the precision of the transfer is enhanced; again, this is the compromise that must be struck between speed and accuracy.

The volume of traffic to be passed, the importance of the information against the time it takes to send, and the prevailing radio propagation conditions will all influence the selection of the most appropriate data mode. Therefore, those that achieve the best data rates, with the most effective error correction systems while remaining confined to a realistic bandwidth, are likely to be looked upon most favourably by the emergency communicator.

Although not the object of this report, it is clear that the most successful system would be one that made every effort to minimise the amount of data that it is totally necessary to transmit. For example, consider a formatted template form where information is entered into a few empty boxes. Virtually the whole file is made up of the template; only a small portion is the unique data. It is pointless and time-consuming to send the whole form. If both ends have exactly the same template, only the parts that are pertinent to the specific record need be sent so long as it is possible to reconstruct the complete form at the receiving end prior to forwarding. This reduces the amount of information that needs to be sent via the restricted link and this could be enhanced further if the data is firstly compressed (The well-known 'ZIP' program is just one example of such a method).

3. Weak signal performance

How well does the mode work when signal strengths are marginal?

Due to the use of correlation techniques and the remarkable sensitivity of the receiving software, data mode signals can be detected and properly decoded even though they are well below the steady-state noise level. All of the modes being compared here are particularly suited to a weak signal environment meaning that low transmission powers can be used. Narrower bandwidth modes tend to be the best performers in this respect.

4. Tolerance to interference

To what degrees do the many vagaries of HF propagation and man-made interference erode the performance of the data mode? How intelligently does the mode cope with such situations?

Short-range VHF/UHF links operate under relatively stable conditions that tend not to change with time. This is not true of HF communications. From one moment to the next, many effects are influencing the wanted signal - both individually and in any number of interacting combinations. These are short-term unpredictable effects and are of concern here. There are also longer-term factors that are more predictable, such as the transition from day to night, seasonal variations and the eleven-year solar cycle. These influences on propagation follow a pattern and can therefore be allowed for in a predetermined communications plan.

It is perhaps interesting to consider that the emergency communicator is very interested in minimising interference caused to their ability to accurately receive data. However, they are likely to be far less interested in their own transmissions being the cause of interference to others – the important axiom is that the information gets through! It would clearly not be the aim to cause deliberate interference, but the choice of data mode is likely to be the one that can survive the electronic mauling of a congested HF band, even if the mode itself is invasive to other users. Ideally of course, a guaranteed clear channel for this activity would be the best option, whether this were through an exclusive allocation or careful negotiation and pre-arrangement with the amateur fraternity at large. It is perhaps important to observe that deliberate interference to emergency communications on a dedicated channel is much more likely to enjoy the immediate attention of the authorities as opposed to the self-policing policy they generally apply towards problems within amateur bands.

Signal strengths

When transmitting data, signal strengths can be remarkably low. This is because the *Digital Signal Processing* (DSP) within the computer software can discern the small variations that represent the information even when the wanted signal is well below the general noise. The threshold at which each mode starts to falter varies.

Under almost all HF propagation conditions there will be a degree of signal strength variation. This may be very slow (*fading*) or quite rapid (*flutter*) and can vary in depth (the difference between the highest and lowest signal strength). Over long distances, when a sky-wave signal path exists between transmitter and receiver, strengths will change as the reflective layers of the ionosphere vary in their position and electrical make-up. However, it is also likely that more than one reflection will occur simultaneously so that constructive and destructive interference is caused at the receiving location.

Selective fading

The major problem for data reception is that there can be small time-of-arrival differences between concurrent paths leading to an effect known as *selective fading*, where individual frequencies cancel across the audio spectrum in a 'sweeping' sound well known to HF broadcast listeners [see **Figure 4**]. This can halt reception when the wanted frequencies are cancelled out.

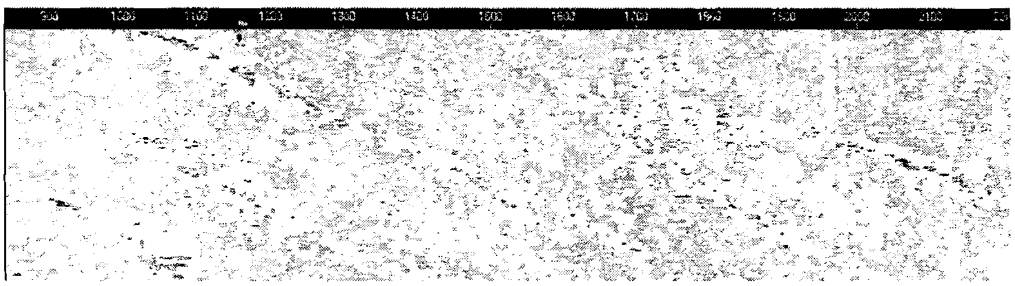

Figure 4: Selective Fading

This is an actual waterfall display of a commercial data station in the 2MHz range. The diagonal lines show frequencies at which there is cancellation, sweeping downwards across the audio spectrum in time and giving the characteristic 'flanging' sound to the signal. The vertical lines are the data signal itself. Modes that send the data more than once over time, or offer the same data in more than one place across the audio spectrum, do well under these conditions.

Delay effects

At very short ranges, the direct ground-wave path can be very stable but only if it exists in isolation (such as 1.8MHz during daylight hours). However, over medium distances it is often the case that sky- and ground-wave paths will coexist (an effect known as *multi-path*), and the difference in arrival time of the two signals can be anywhere between 5ms and 20ms. This can have a serious effect on the synchronisation of data.

When the ionosphere is very disturbed, such as when subjected to major solar activity, violent changes can bring about the Doppler effect. This in turn can cause phase modulation. Even if the changes are very small, this will badly affect those modes that rely on phase coherence. It is particularly troublesome to the slower modes.

Noise

HF circuits can be subject to random noise. There are several categories of noise, all of which can affect the performance of data modes.

Steady *broadband noise*, due to natural solar effects, adds to the hiss inherent in all of the electrical circuits of the equipment being used. Man-made radiation, such as from domestic equipment, industrial plant and the electricity supply network, also provide a constant background noise level. One problem connected with broadband noise is that it can mask the very weak data signals, making it hard to tune the receiver to the correct frequency on visual 'waterfall' displays. In general, narrow-band modes tend to perform better in broadband noise conditions.

Noise that is only present from time to time is known as *impulse noise*. The most prevalent natural source of this at HF frequencies is lightning discharge. The same attributes of the ionosphere that allow signals to travel long distances mean that the 'static crashes' from lightning can also be heard from all over the planet.

During a burst of lightning noise, a data signal may be completely overwhelmed for a brief period. Therefore, the best modes to use are those that can keep synchronisation during a momentary disappearance of the signal. Another key issue is how quickly a data decoder can re-lock to the data bitstream after it has been temporarily masked, in order to minimise the loss of information. Lightning impulses can also produce very high signal readings in radio receivers. It is therefore important that the AGC of the receiver is appropriately set, so that it is desensitised for the minimum amount of time.

Man-made sources of impulse noise include vehicle ignitions, arcing thermostats, power tools and certain broad-spectrum military transmissions.

Many modern radio receivers have special facilities to reduce both wide-band and impulse noise. However, as outlined in **Appendix A**, some data signals can appear remarkably like noise to these circuits. Very careful use of these facilities is therefore recommended, otherwise the very signal required might be digitally erased!

5. Efficiency (RF bandwidth and Duty Cycle)

Different data modes modulate the transmitter in very different ways.

Spectrum efficiency

One area of efficiency that needs to be considered is the economic use of radio frequency bandwidth. This may or may not be an issue, depending on the status of the channel being used. Both from a social point of view as well as for limiting the possibility of interference from other users, the less space taken up the better within a busy amateur band. Conversely, if a dedicated channel is in use, the bandwidth taken up should *maximise* the allocated segment. As a general rule, the more bandwidth used the greater the data rate and the more error correction can be redundantly applied.

Average transmitted power

Another area of efficiency is the demand made upon the radio transmitter. The *duty cycle* of the equipment refers to the ratio of receive, stand-by [3] and transmit. In broadcast applications, the system is effectively transmitting all of the time. But this is not the whole story!

Some modes produce constant amplitude, which means that the transmitter is continually generating power output. This requires a more robust amplifier stage than when using modes that have periods when no power is being produced, such as normal voice transmission. Indeed, it is a fact that virtually all data modes place a heavier demand on the transmitter than speech.

It should also be noted that the average power developed by any particular signal is related to the bandwidth used. Therefore, the estimation of average power output is the best way to assess the efficiency of a data mode.

Measuring power output from a radio transmitter when sending data is a distinctly tricky business. Even the top-range RF meters are likely to be set up for measuring speech-like signals. This is the case because the data signal can be very complicated; particularly when multiple tones are used and which are not necessarily present all of the time. Note, for example, that when data is being transmitted there is a marked difference in output reading as the meter movement is switched between 'fast' and 'slow' ballistics. This is because the former is attempting to react to instantaneous peaks while the latter is damped to provide more of an average reading. The only true measure of power output is to generate a single sine-wave tone for setting up the transmitter. Alternatively, some data modes effectively produce this, but mostly *when idling only.*

In practice, the greater the average power, the higher the temperature of the *heatsink*, a large metal flange with vanes that dissipates heat away from the output amplifiers. This is because heat is the principal unwanted by-product of the wanted radio frequency energy, produced as a result of inefficiency in the amplifier components. Over time, the risk of damage to these is clearly greater if the transmitter is being driven close to its upper power limit. Unlike most commercial transmitters, amateur equipment is not usually designed to operate continuously in this way. A cool heatsink is a sign that the transmitter is not being overworked. Perhaps it should be remembered that a doubling of output power is only an improvement of 3dBW, so it will always be best to keep output powers to a minimum.

Excessive output powers also place heavy demands on the supply to the transmitter; this may be an issue if a limited, temporary source of power is being used (such as a generator).

Operating at maximum power output for speech is acceptable because of the large gaps between human vocal sounds, allowing the amplifiers to rest. The transmitter is designed to work continuously under this sort of demand. Data modes generally do not exhibit such gaps and this must be coupled with the fact that in a broadcast station the transmitter will be constantly sending for many hours. Therefore, a good rule of thumb would be to use the minimum drive necessary to guarantee good communications and never more than 50% of the full output power available. Fortunately, as has already been discussed, data modes lend themselves to lower power output requirements than speech due to the excellent sensitivity of the receiving software.

[3] 'Stand-by' is a somewhat historic term these days and reflects the situation that prevailed when equipment tended to be fully transistorised on receive but still used valves to produce the high energies required for transmitting. 'Stand-by' referred to the state where the valves were energised (i.e. power supplied to heaters and plates in preparation for going on air) but not yet actually transmitting. This state would draw more current than simple receive, but not as much as full transmit.

6. Ease of operation

Unlike the more stable environment of the home amateur radio station, operators may not be as acquainted with the set-up encountered in the field. The equipment may not be theirs. Unfamiliar hardware and software need to be quickly mastered. Therefore, the simpler the function and the more tolerant the particular mode is to minor operational errors, the better.

Software issues

It is best to start by admitting that virtually all of the software presently available for operating data modes leaves a lot to be desired in terms of the human interface. The thrust of the program is to control the data mode, often leaving the actual information handling at a fairly rudimentary level. These are matters that could be resolved by writing improved operator interfaces. However, most current packages cope well with the Windows style of 'drag-and-drop' where text from an external file can be highlighted, copied, then pasted into the sending box of the communications program and vice versa. Luckily, this is a comfortable operation for the basic computer user. Conversely, importing and exporting binary files can be a bit cumbersome; yet an understanding of these processes is vital to manipulating the User's information.

Control of the radio

The ease with which different data modes can be used on the air varies considerably. For example, some modes can only be decoded when the receiver is selected to the appropriate sideband; others are not concerned with this restriction.

It is probably the matter of correct tuning to the radio frequency signal that can make all the difference between total success and utter failure. Many data modes occupy minimal bandwidth and use small phase or frequency changes to represent the digital information. To discern these fractional changes, accuracy of tuning is vital. It is equally important that frequency drift, a problem with older equipment, is minimised. This may be achieved by the replacement of components, such as a high stability crystal in the master oscillator.

Those unused to data modes can be initially confused by the fact that it is not always necessary to tune the radio critically. This is because the software examines the whole of the speech bandwidth available to it, some 3000Hz or more, in which there could be as many as thirty separate narrow-band data transmissions happily existing side-by-side. So to determine which bitstream to focus the decoding software upon, the operator tunes the *computer* up and down the audio spectrum rather than the radio. To aid the operator, most modes include an audio AFC (automatic frequency control). This allows the program to track any small changes in the carrier frequency of the signal and to home in the decoder when it is not quite correctly tuned. However, the radio/computer combination must be fairly accurately tuned to the wanted bitstream to begin this process. Once this has been achieved, the AFC should compensate for any slight drift of the signal.

A problem often arises because the strength of the desired signal is so low in level that it can be very difficult for the operator to identify it on the computer waterfall display in amongst all the unwanted signals and noise. Some modes require precise tuning before any meaningful data is decoded; others are quite tolerant of tuning inaccuracy. In terms of getting going, this is important to the operator's enthusiasm!

It is also to be noted that modes which employ time-based *interleaving* may be being received quite perfectly yet the resolved text may take many seconds to appear on the screen. Receiving operators need to be disciplined in waiting for a sufficient period otherwise there is a danger of making tuning adjustments just as the required data starts to emerge.

Some of the wider bandwidth multiple-tone modes are particularly difficult for the human to interpret. Not only do they sound like noise, they even look like noise when examined in both the time and frequency domain! At low signal strengths, there is no discernable difference between the signal and the noise. To an extent, this problem has been worked around by agreeing accurate carrier frequencies for operation.

7. Compatibility and adaptability

Are there compatibility issues when the sending and receiving ends use different emulations of the same data mode? How possible is it to adapt (or even modify) software to meet the needs of emergency communications?

Different versions of software

Although a single person or group will have developed the original software code, for any given data mode it is likely that there is more than one computer program that is licensed to include it. Where this is the case, there may be differences in the operational presentation of the mode. Although the fundamental encoding and decoding routines are likely to be fixed and protected from modification by copyright, other features may or may not be included in the new emulation. For example, MT63 has the facility for a secondary channel running simultaneously alongside the main channel. This can be put to a variety of uses, such as the generation of a continuous identification or *beacon*. However, this is not a prime function of the mode and therefore some software provides for it and others do not. The option to transfer binary files, such as higher-level documents or spreadsheets, is similarly at the whim of the programmer.

Although there are key parameters, such as bandwidth or baud rate, that are always user-adjustable, the more comprehensive programs may allow more fundamental values to be altered. Although this improves flexibility, it can also provide a trap for the less well informed. Changing these values has the potential to render the data mode unworkable and there is not always a simple method of reverting back to the 'standard' settings. It is therefore important that the unwary are taken in to account by the programmer should they stray in to these areas of the configuration.

For example, some programs offer the option to alter the master synchronisation clock of the program. Of course, it is vital that this is treated with extreme caution. Likewise, some software offers an interesting selection of soundcard-based DSP filters. It is very unlikely that any adjustments to these will improve reception.

As has been previously discussed, the editing and file handling facilities of much of the present software are basic to say the least. It is interesting to note how different the operator's impression of the data mode can be, depending on the quality of the interface they see and have to work with on the screen.

Software adaptation

Adapting software to suit the needs of emergency communications is a challenge that has yet to be mounted. Many of the packages presently available have a 'contest configuration' that changes the priority of the screen and keyboard function away from casual amateur use, to make operations as quick and easy as possible for the contest enthusiast. At the speed of contest operation, factors such as rapid logging and the provision of single-key shortcuts to pre-programmed text macros become very much more important and are therefore enhanced.

Regretfully, an 'emergency configuration' has not so far appeared. One could envisage that this might include improved file handling, a message precedence regime and tracking logs. It remains a matter of convincing the programmers that embarking on such a redesign is worthwhile.

Alternatively, it might be more useful to create a separate program that does all the things that emergency communicators require, but then automatically exports the material for transmission to a standard data package in the form of a 'task list'. Using a separate preparation program would have the advantage that message preparation could take place at any time, whether or not the communications program was running.

8. Reliability

When great reliance is placed in the chosen software it is important that it does not 'crash' and that it runs well on a variety of computers.

Software for data communications should configure easily and should not place overdue demands on the resources of the computer. Although PCs are becoming much more powerful day-by-day, some of the more complicated data modes can still stretch a modest machine. This is particularly true if other software is simultaneously running on the one computer.

The configuration settings of the data program can radically affect how much processing time it claims of the PC. For example, most data programs show the received signals as a *waterfall* display. This is a continuous, and often colourful, rolling image of the audio spectrum that allows the operator to guide the decoder to the desired signal. It is vital to successful operation. However, it is also usually possible to alter the size of this window and should it be made too large this can cause the computer to act unpredictably as it struggles to create the picture. The speed of the waterfall display should also be kept to a minimum, for the same reason.

Another disturbing result of making too great a demand on the computer is that the transmission or reception of data is momentarily halted. When transmitting, the effect can be a break in the bitstream caused by the processor becoming overwhelmed by another task – the saving or opening of a file for example. This could be catastrophic as the brief loss of synchronisation at all of the receivers could cause data to be misread. To give the machine the best chance of performing at its optimum, unwanted programs that are running in the background should be closed down.

9. Availability

Is the software easy to obtain?

Luckily, most of the data programs can be downloaded from the Internet without cost. This is more generally true of the smaller, more basic offerings that tend to concentrate on a few modes only. The bigger packages, offering all of the popular data modes side by side in one suite, tend to require a small registration fee. This really is a small price to pay for the benefits received. Using the more comprehensive programs gives the user total flexibility to jump quickly from one mode to another within an integrated environment where many of the useful user-set features carry across.

This having been said, there are advantages to using a smaller program. Concentrating on a single mode is likely to tailor the whole presentation to the most suitable configuration and the demands on the computer processor are likely to be less.

Without doubt, the wholesale introduction of a particular mode as being the most suitable for emergency communications is unfortunately likely to depend upon minimal cost as much as its technical merits. If the software needs to be widely distributed, suggesting a program that is not free of charge could thwart progress. This could therefore exclude some of the bigger packages, which may be unfortunate when it comes to ultimate flexibility in the field.

RESULTS OF EXPERIMENTS

To support the on-air testing already described, desk-top experiments were also set up. These used the CoolEdit Pro program (see **Appendix B**), a professional audio editing package used by broadcasters for the preparation of sound files. This facility has a *multitrack* window in which individual sound files can be mixed together. For example, by using this feature it was possible to add background noise to a clean data signal.

Great care was taken to ensure that the original data tracks were distortion-free. Then, through use of the CoolEdit parameters, it was possible to deliberately modify the audio waveform to introduce effects that might be typical of a real installation. For example, it was possible to apply small delays to one signal then add this back to the original waveform in varying ratios, thus emulating a multi-path effect. It was also possible to cause shifts in pitch and to superimpose several data signals on top of one other.

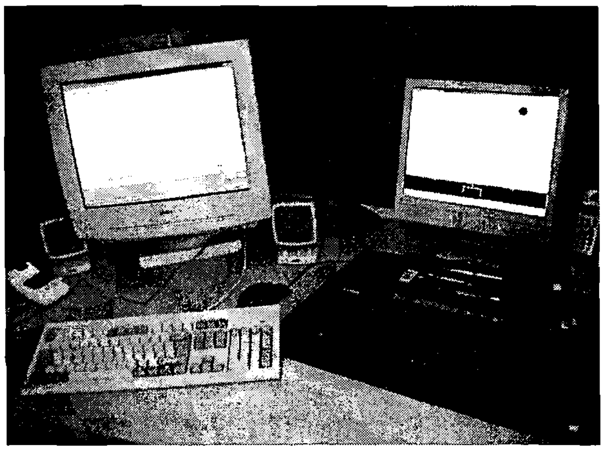

Figure 5: Computers testing the data modes

The author's test facility. The soundcards of the two PCs have been joined together. IZ8BLY MT63 is seen on the left-hand screen, while MIXW2.10 MFSK16 is running on the right-hand machine.

As well as from sources already within the software, such as *white noise* or *sine-wave tones*, several actual recordings from HF receivers were also made. These provided realistic tracks of background noise, impulse noise such as lightning, and splattering SSB speech! For many of the desk-top tests, the soundcards of two PCs were joined together [see **Figure 5**].

1. Accuracy (error correction)

BPSK31 and QPSK31

BPSK31 uses no error correction, but this mode champions the advantage of small bandwidth. The later development of QPSK31, with four phase states to play with, allowed the addition of *convolutional* code error correction. This technique uses extra information to describe a past history of the data, allowing some reconstruction to occur. Unfortunately, QPSK31 is also more susceptible to phase errors, something common over HF paths, thus this somewhat reduces the impact of the improved error correction. Comparing QPSK to BPSK over more stable paths, however, shows QPSK to be a better performer.

MFSK16

MFSK modes also use a convolutional code error correction for strong FEC. Time interleaving is also employed. Together they make this mode about the most sensitive receiver available. Perfect copy is possible deep into the noise.

MT63

MT63 is perhaps the most elaborate user of error correction techniques. It uses a Walsh function that spreads the data bits of each character across all 64 of the tones of the signal spectrum and simultaneously repeats the information over a period of 64 symbols within any one tone. This takes 6.4 seconds. The combination results in superb impulse noise rejection. At the same time, in the frequency domain, significant portions of the signal can be masked by unwanted noise or other transmissions without any noticeable effect on successful reception. Transmission speed is good because there are so many individual tones to describe the information, while the individual symbol rate per tone can remain slow (which is good protection against ionospheric disturbances).

Binary files

The IZ8BLY version of MT63 allows the attachment of binary files, but MIXW only allows the importation of text files. The MFSK16 routine of MIXW will allow pictures to be sent and received.

2. Data rate

BPSK31 and QPSK31

Both the bi-phase and quad-phase versions of PSK31 are similar in many respects. They both use a *varicode* encoding technique, meaning that the length of the bit-word that describes each ASCII character is not the same. The most popular characters (such as 'space' and the letters 'e' and 'o') have much shorter codes than the characters that are used more infrequently. This does mean that the data throughput is faster than might have been predicted but is still not particularly rapid.

MFSK16

As has been stated, this mode uses a combination of interleaving and convolutional coding. The application of these error correction systems approximately halves the potential data rate. It must also be stated that the interleaving exercise results in a six-second delay between reception and the decoded text appearing on the screen.

MT63

The much greater bandwidth of MT63 permits high data rates to be achieved. However, as detailed in the EASE OF OPERATION section below, the amount of error correction leads to significant delays between the signal being received and the data being decoded. This can lead to confusion and frustrated adjustment of the equipment, even though there is actually nothing wrong with the system!

MT63 at 2kHz bandwidth is extremely fast and really does not lend itself to manual typing at the keyboard. At this setting, text should be prepared in advance and then 'cut-and-pasted' into the transmission window. The speed performance of MT63-2K is also much more practical for the attachment of binary files.

Table 1 summarises the findings. It is based on the many occasions when the Radio Society of Great Britain's GB2RS new bulletin has been transmitted on 1.8MHz and 5.0MHz by the author, using all of the modes of interest here.

	SPEECH	RTTY 50 baud	QPSK31	BPSK31	MFSK16	MT63 1K	MT63 2K
Transmit time (mins) [a]	40	76	76	76	67	41	19
Symbol rate (baud)	--	50.000	31.250	31.250	15.625	10.000	20.000
Words per min.	100 approx.	66	35	35	40	100	200
Delay RX-TX (secs) [b]	0	1.5	2.0	1.5	6.0	12.0	6.0

Table 1: Data Rate Comparison

[a] This is the time taken to transmit a typical GB2RS news script, including the main news, propagation update and local items.

[b] This is the time that elapses, under typical conditions, between a character being typed at the sending end and it appearing on the screen at the receiver.

3. Weak signal performance

Steady-state noise test

To explore the relative performance of the modes, a clean test transmission was recorded for each mode. This consisted on a ten-second idle period, followed by the transmission of the following text:

"When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity."

Using CoolEdit Pro v2, each mode in turn was 'transmitted' to a second computer on which data-decoding software was running, but with white noise added. The signal level of the wanted data signal was gradually reduced until the text started to garble, continuing until the original text was unrecognisable. At all times, great care was taken to eliminate any phase or level distortion during the tests.

Table 2 shows the results. The green portions of the table indicate perfect reception. The yellow portions represent text containing some errors. Text that is too garbled to be recovered by human interpretation is coloured red.

Audio S/N ratio	MT63-2K	MT63-1K	BPSK31	QPSK31	MFSK16
-8 dB					
-9dB	When trEnsmitting data it is vital to set up each point in the audio chain to prevent ditortion and non-linearity.				
-10dB	;9||E`xH3nx.mQj/[J~=LZ(UJZt>BgY1n4R= ~1bD'!^tJwrVd; %W~en tr|nsmi[ing 0a[aFit				
-11dB		When transmittingvital to set up each point in the audio chain to prevent distortion and non-linearity.			
-12 dB		When transmittingdata it is vital to set upDeach point in the audio chain to prevent distortion and non-linearity.	When transmitting data it is vital to set up each point inthe audio chain to prevent distortion and non-linearity.		
-13 dB		When+transmittngdata it is vital to set upDelch point in the audio chain to prLvent distortion and non-linearity.	When transmitting data it is vital to set up each point n the audio chain to prevent ditortion and non-linearity.		
-14 dB		}kF7(U-XZtki6+is}ZiSallo set uptlc int intha audio chaiQ o[zprf3t H3slKdnGB> fC\K	When transmitting data it is val to p each point e audito prevent ditortion an non-linearity.		
-15 dB			When smittig data it is vital to set uph poinin the hain to prevent drtion ad non-linear o	When transmitting data it is vital to tet up each point in the audio chain to prevent distortion and non-lineaarty.	
-16 dB			mittig datas vita to prevent dineari o	Wt n transmitting datrft is vi to set up each point in the audio chain to prevent distortion and non-linear ty.	
-17 dB			al to so prent di	When transmitting data it is vital to set up]nh poiat in the ama o cha to prevenrfstortion ant non-f.ear ty.ne	
-18 dB			a it isrevent	]Aew6 Wh tranitfting datrrt sn vital et ak ea æ pointAn the roa (chaio to prevent ütort]n ana noe rloerity.	
-19 dB				ttte tmeo ing data=t irta rtet sp Ysh eAheá‹ o chain to pr tpnt distortion aor btineact (t	¢hen transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearitetR
-20 dB					Wh transmitting datt t ueach poitnthe udio chain to prevent distortioand olinearitÊ
-21 dB					i p

Table 2: Weak Signal Performance

Notes to Table 2:

The signal-to-noise ratio given indicates the level of the wanted signal below the noise level. This relates to the full audio spectrum and therefore does not necessarily correlate with the signal-to-noise reading given by the data mode program, which is only measuring the ratio within the bandwidth of the mode in use.

For the purposes of this test, the data squelch facility of the receiver was switched out. MT63 displayed some interesting characteristics in this situation. Firstly, random characters would appear on the screen that appeared to be an attempt to decode the white noise alone. However, as soon as an idle signal was detected, the random character generation ceased. This splurge of unwanted characters at the beginning of the transmission has been removed from the table for the sake of brevity.

At very marginal signal strengths, it appeared that a suitable setting of the MT63 receiver data squelch threshold could not be found. In this situation, it was possible to decode quite accurate text that would otherwise have been muted by the use of the squelch. The disadvantage of operating like this would be that whenever the signal truly dropped into the noise, beyond any chance of decoding, a significant number of random characters would be printed.

It is interesting to note that MFSK16 performed exceptionally under these conditions [see **Figure 6**]. Even more impressive, perhaps, was the fact that good copy was maintained down to a very low signal level, then, within a decibel, the receiver printed virtually nothing. This is in stark contrast to the performance of MT63 at these marginal levels, as described above.

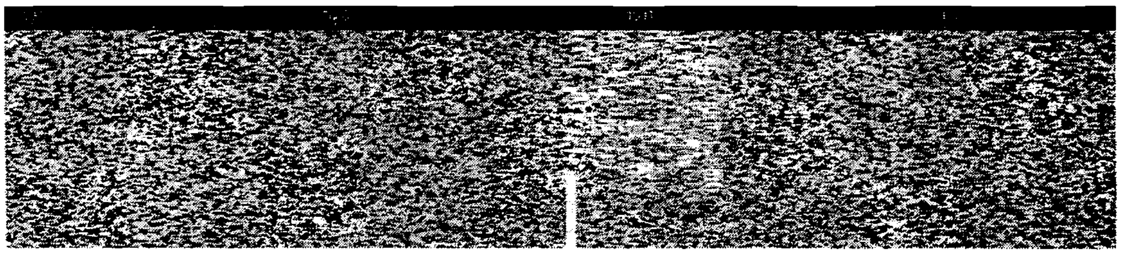

Figure 6: MFSK16 and noise

Note how difficult it is to identify the signal once the idle tone disappears as data transmission commences. This is a disadvantage of using MFSK16.

At marginal signal levels, it was noted that both PSK31 variants would produce random characters that appeared to be an attempt to decode the random noise.

The weak signal tests were also extensively carried out on the air. The same short message was transmitted in quick succession to a number of stations located all over the United Kingdom. With each transmission, the power output used was reduced by 3dB. The results married well with those shown in Figure 3.

4. Tolerance to interference

Impulse noise test

This test was similar to the steady-state white noise test above, but this time the noise source was of the impulse type. To create this, HF receiver noise was recorded with the author generating random bursts of noise, some short, some long, using an electric drill nearby. The result was a reusable track of band noise plus short static bursts not dissimilar to lightning.

The relative levels of the steady-state receiver noise against the wanted data signal were set to the lower threshold of perfect copy. The impulse noise was therefore an occasional additional burst that was likely to mask the wanted signal.

BPSK31 and QPSK31

The minimal bandwidth of these modes appeared to be an advantage, because the wideband noise made relatively little impact across the spectrum of the wanted signal. The bursts of noise tended to produce unwanted characters. Longer bursts were more disruptive than short stabs. Errors in reception coincided closely in time with the noise bursts, but both modes recovered almost immediately after the noise had gone. This was probably due to the lack of error correction in BPSK31 meaning that it was making no pre- or post-noise error calculations, although the performance of the quad-phase version was only vaguely better.

MFSK16

The time-based interleaving of this mode coupled with the noise-rejecting small bandwidth allowed MFSK16 to out-perform all of the other modes. Reception was highly immune to short bursts of noise. However, the longer bursts (there were a few lasting approximately three seconds) took their toll and it was noticed that MFSK16 was slow to recover in this situation, taking typically two seconds to resynchronise. As previously, it was interesting to note that the difference in signal strength between perfect copy and virtually unreadable text was just 1dB!

MT63-1K and MT63-2K

Due to the time-based interleaving, both modes fared well against even the longest bursts of noise. Interestingly, despite utilising double the bandwidth (and therefore theoretically being more disadvantaged by the broadband noise bursts) MT63-2K performed better than the 1kHz version. Perhaps this was due to the higher baud rate, allowing more data and error correction bits to be passed in the gaps between static bursts.

It was obvious that with both versions there was considerable error correction being used. This manifested itself in the very long delays experienced between the signal being received and the text appearing on the screen.

It was noted that there tended to be a gap in copy rather than a significant generation of unwanted characters at the point of a sudden noise burst. Perhaps this was because the receiving modem would have been fully synchronised up until the start of the burst. This was in contrast to the effect seen when the MT63 decoder was exposed over a period of time to random white noise only, when a large number of unwanted hieroglyphics were seen to appear.

Common to all modes was the observation that the longer the wanted signal was not present, the more data would be missed. It is worth restating that the need for a fast AGC recovery on the receiver during impulse noise conditions leads to more complete reception.

PSK31 at –24dB

"When transmitting data it is vital t set u enc=i< in œe audio ctaaAto prevent dintortion and non-lineare yr"

QPSK31 at –19dB

"When transmit rng data it is vital taEet ea- point in the audio chaia to prevent distorOel and f n-linearittto"

MFSK16 at –25dB

"When transmitting data it is vital to se¡ point in the audio chaiuxrevent distortion and non-linearity."

MT63-1K at –12dB

"When transmitting data it is vital to setDup njc5 point in the audstortion and non-linearity."

MT63-2K at –14dB

"Whentransmittirg Data it i set up each boint6in the audio chain to prevent dist;rtion Anl nWn-lin]"

Impulse noise performance

Selective Fading test

In order to provide precisely the same conditions to apply to each mode, CoolEdit Pro was used to generate a mixture of white noise and the wanted data signal under test. The relative levels were then set so that the lowest threshold of perfect copy was found. Then, each waveform was individually subjected to a flanging filter (usefully called "Short Wave Radio" by the software!) with the parameters set to mimic the sweeping cancellation characteristic across the spectrum known as *selective fading* [see **Figure 7**].

Flanged White Noise parameters:

Sweep Period - 3s

Delayed signal - delay 5ms

Mixture - 60% original, 40% delayed

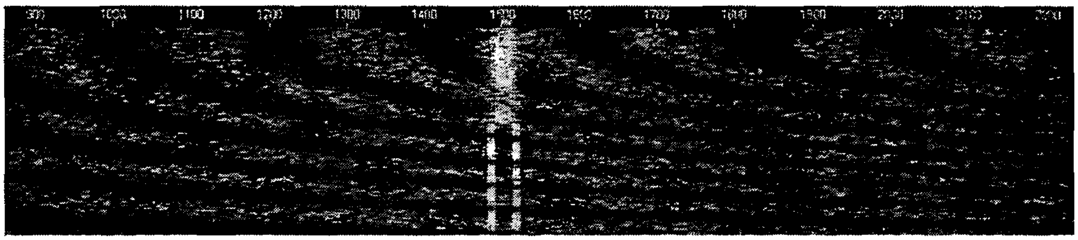

Figure 7: BPSK31 and noise subjected to the "Short Wave" flanger

The black diagonal bars are the frequencies at which cancellation is occurring. It can be seen that as the sections of cancellation sweep downwards is frequency through 1500Hz there is noticeable destruction of the PSK waveform.

BPSK31 and QPSK31

Both versions of PSK31 suffered badly from the cancellation effects of the selective fading. The small bandwidth worked against these modes because there was effectively almost total removal of the signal at times, punching significant holes in the bitstream. With no interleaving to rescue the situation, the result was catastrophic, reducing perfect copy to virtual unintelligibility! However, the ability of these modes to recover quickly outstrips the others.

MFSK16

This mode was fairly immune to the selective fading but in the situations where the frequency sweep was moving rapidly through the signal spectrum there was a loss of data. Significantly, MFSK16 took a lot longer than the PSK31 modes to recover from this situation. However, it should be noted that this mode was still successfully working in the selective fading environment at *very* much lower signal levels than any of the other modes under test.

MT63-1K and MT63-2K

The result for these modes conclusively demonstrates the advantages of spectral and time-based interleaving working together to recover from a fairly harsh attack by the ionosphere. In fact, for MT63-1K, only a single misplaced character was induced which is remarkable considering that the signal was virtually undetectable on the waterfall display [see **Figure 8**].

Figure 8: MT63-1K and selective fading

The MT63 signal can hardly be seen on the waterfall display, yet sill gave virtually perfect copy.

BPSK31:
"When ansmitting data it is vttal to set uc each point in ehe audio chaia to prev nt distortionhnee ooinearity."

QPSK31:
"Whransmitting data it ih l to set e each rnt in the al chaanto preoe t distortiearJ ty."

MFSK16:
"When transmig data it is vital to set up each point in the audio chain to prevent distortion and non-gÐ tÜ01Po1t"

MT63-1K:
"When transmitting data it is vital to set up each point in the audio chaiE to prevent distortion and non-linearity."

MT63-2K:
"When transmiating data it is vital to set up each poijt in the audio chain o prevent distortion and non-linearity."

Selective fading performance

Delay/multi-path test

For this test, each mode was mixed with a delayed copy of itself. The idea was to recreate the condition often found over medium-distance paths whereby the ground-wave and sky-wave both reach the receiver, but with the latter taking longer to arrive over the refractive path via the F2 layer. Taking into account the possibility of NVIS sky-wave reflection (as previously outlined), the range of typical delays that might be encountered is 5ms-20ms.

Clearly, these tests only touch on the complexity of the true situation. In practice, there may be many paths by which the same source signal is arriving, and each with its own relative level. While these tests have only made at intervals of 1ms, mathematical delays at which total constructive or destructive interference occurs have not been specifically examined. It is hoped, however, that the following results at least hint at some of the complex issues at foot when examining these effects.

PSK31 modes

The worst-case situation for BPSK31 would be a delay of 32ms with both-ground wave and sky-waves being of equal magnitude. The period of the raised-cosine carrier pulse is 32ms and therefore something approaching total cancellation would occur. However, this amount of delay is very unlikely. QPSK31 has envelopes of different periods and therefore this issue does not arise.

BPSK31	DATA RECEIVED	COMMENT
5ms	o eon o pe epen o o oeo e oon eno nono ne neo e e n n	Noticeable cancellation of waveform and total loss of data
6ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Strong constructive interference
7ms	n eeen e e ennoeo ee noe o t o e n neo t e eo oee o	Severe waveform cancellation
8ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Constructive interference
9ms	n e t t e o e o i e et t nee e o oe e ee e oo o e	Severe waveform cancellation
10ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Strong constructive interference
11ms	e t o t tel oet t oueo-tion oni nono i neo'tyc	Severe waveform cancellation
12ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Strong constructive interference
13ms	nei tranne o eo to tt t ieo t e tee en o ne e e eo o to	Severe waveform cancellation
14ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Strong constructive interference
15ms	raoae tting t atd t ia o tae te e p each pe ot n the afdio tta e revet t iatertiee e non-i e ritys	Fairly strong destructive interference
16ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Strong constructive interference
17ms	hen tranom .s e ia a it io oital h et up ench point io the aud7retin h ei ot t istortieo os eon- eiee rit=ô	Fairly strong destructive interference
18ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Strong constructive interference
19ms	ei,oe tae ua e cie e Û we, io or ieo andsoe-oio aret e s	Severe waveform cancellation
20ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Strong constructive interference

BPSK31 Multi-path performance

It is perhaps interesting to note that when the two interfering signals are different in level by more than 3dB the cancellation effects that are otherwise so noticeable (such as at 7ms) are almost dismissible. This being the case, the actual on-air experience (with the relative level of the sky-wave portion constantly changing) is likely to be cancellation for a few seconds only.

At certain delay lengths, one or other of the characteristic 'tram lines' of the idling PSK31 waveform can be completely cancelled. However, it was found that this did not necessarily mean that data reception was completely lost.

It is also useful to note that PSK31 prints little to the screen under these circumstances. This means at least that the wanted text, when received correctly, is not masked by a large amount of unwanted 'hash'.

MFSK16

The results for this mode were really quite remarkable. The principle effect noticed was that cancellation of specific tones within the sixteen was occurring. However, it was apparent that quite a few of the tones could be missing (perhaps five or more) but that the data could still be recovered correctly. Indeed, there was only one setting of the delay that caused any loss of data and this may have been due to the loss of both the key lower idle tone, the highest tone plus three other bands of cancellation in-between. This shows the benefits of the redundancy in the error correction system. Even then, if the delayed sky-wave signal fell by as little as 3dB below the ground-wave signal, then copy was restored after a longish period of resynchronisation.

It should be noted that the loss or deep suppression of the most extreme idle tones did cause problems, however. In time, there was a tendency for the decoding software to drift without these references. Nevertheless, as a transitory effect, MFSK16 is not going to be all that bothered by multi-path issues.

MFSK16	DATA RECEIVED	COMMENT
5ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Noticeable cancellation of middle tones
6ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Noticeable cancellation of some upper and lower tones
7ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Noticeable cancellation of middle tones
8ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Noticeable cancellation of some upper and lower tones
9ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Noticeable cancellation of extreme upper and lower tones (including idle tone)
10ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Noticeable cancellation of middle tones
11ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Noticeable cancellation of middle and some upper and lower tones
12ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Four noticeable bands of cancellation including extreme upper and lower tones
13ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Noticeable cancellation of middle and some upper and lower tones
14ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Four noticeable bands of cancellation
15ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Noticeable cancellation of middle and some upper and lower tones
16ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Four noticeable bands of cancellation
17ms	ytmitting ta it is vital to set up each point i-vAdl...hain to G'P distohtion ue˜ ta-6llofZ	Five noticeable bands of cancellation including extreme upper and lower tones
18ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Four noticeable bands of cancellation
19ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Five noticeable bands of cancellation
20ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity	Four noticeable bands of cancellation

MFSK16 Multi-path performance

These very good results are largely to do with the slow symbol rate for any individual tone. The mechanism employed in MFSK16 is to ignore the leading and trailing 5ms of each tone in order to dismiss time-of-arrival differences that would otherwise cause interfering overlaps. In theory, delays of at least 10ms can be accommodated but, as shown above, in practice the tolerance to multi-path effects is dramatic. It should be noted that the slow individual symbol rate does not mean that the data rate itself is also slow, an advantage of using a sixteen tone system.

MT63 modes

With such a wider audio bandwidth than most other modes, the effects of applying delay to MT63 are very apparent. The 'comb filter' effect can clearly be seen on the waterfall display in **Figure 9**.

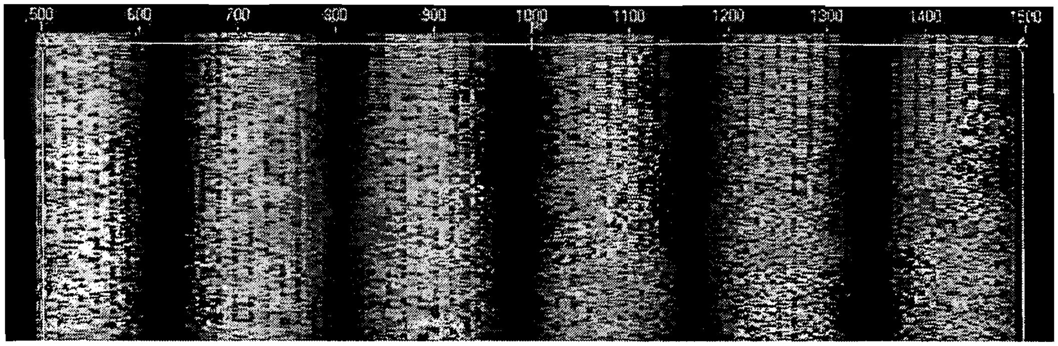

Figure 9: MT63 multi-path reception

Note the very obvious bands of signal cancellation spread across the spectrum.

In a practical situation, the cancellation bands would probably not remain static as in **Figure 9**, but would move about as the sky-wave time-of-arrival varied with the vagaries of the ionosphere.

As can be seen from the results below, MT63 is not troubled by the cancellation of even 50% of the 64 tones. This is a good example of the interleaving that is applied across the audio spectrum. If the data is temporarily or permanently unavailable at one point on the spectrum, it can be found at several others. This accounts for the amazing performance of MT63 in a multi-path environment. It generally remains error-free even under the most trying of conditions.

MT63-1K	DATA RECEIVED	COMMENT
5ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Five bands of cancellation
6ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Six bands of cancellation
7ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Eight bands of cancellation
8ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Eight bands of cancellation
9ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Ten bands of cancellation
10ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Ten bands of cancellation
11ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Twelve bands of cancellation
12ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Twelve bands of cancellation
13ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Fourteen bands of cancellation
14ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Fourteen bands of cancellation
15ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Sixteen bands of cancellation
16ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Sixteen bands of cancellation
17ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Eighteen bands of cancellation
18ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Eighteen bands of cancellation
19ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Twenty bands of cancellation
20ms	When transmitting data it is vital to set up each point in the audio chain to prevent distortion and non-linearity.	Twenty bands of cancellation

MT63 Multi-path performance

Tolerance to nearby signals test

In this test, the signal spectrum of the data mode in question was deliberately invaded by unwanted transmissions. The interfering signal could be virtually any kind of transmission: a constant carrier, speech, CW, or a data mode. This might occur when the sender of the interfering signal had not detected that there was a weak data signal already present on their choice of frequency. This is common on the amateur bands where some users do not recognise the 'noise' they are hearing, or choose to ignore it.

Interestingly, even dedicated channels, such as those currently afforded to amateurs on 5MHz, are not immune from problems. Firstly, there is a strict priority amongst the potential users of such channels and a military station is entirely within its rights to commence transmissions regardless of any signals already present. There is also the distinct possibility of strong AM carriers appearing after dark when conditions favour long-distance signals. This can occur due to international differences in band-planning, meaning that a legitimate broadcaster from a far-off country can unwittingly cause problems to local communications within the UK.

PSK modes

The small bandwidth of the PSK modes had several advantages. Principal amongst these was the number of separate transmissions that can exist in close proximity to each other. In **Figure 10**, two separate transmissions of equal signal strength were extremely close to each other, yet perfect copy was received from both.

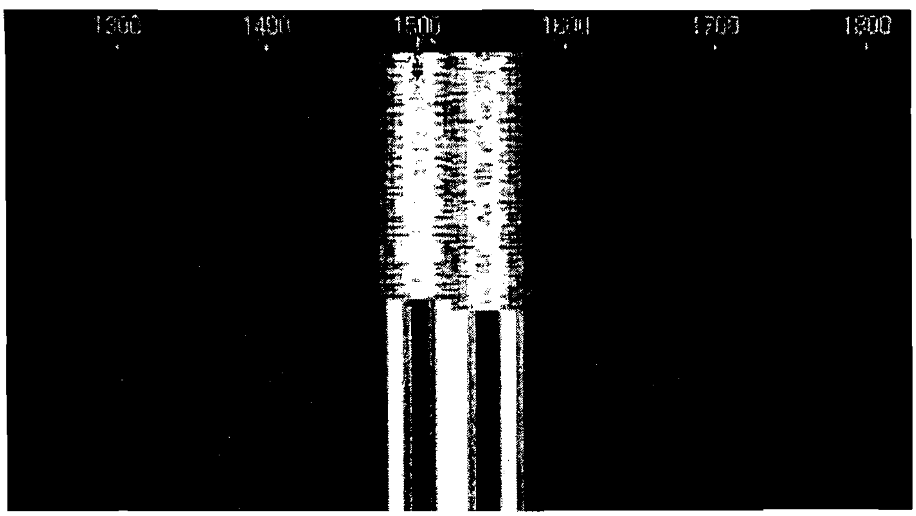

Figure 10: Two adjacent BPSK31 transmissions

The two signals very nearly overlap, but not quite. Perfect data was received from both signals.

This example probably showed the absolute limit of acceptability in terms of coexistence. Once the two signals overlapped, total destruction of the data occurred even when the difference in signal strengths was significant. The same was true of interference from single sine wave tones such as a constant carrier or CW transmission. PSK31 could work right next to these signals but was badly affected as soon as there was any overlap.

It should be observed, however, that the small bandwidth of the PSK31 modes means that about thirty separate signals can be deployed within the same bandwidth as a single speech channel. The possibility of sending the same information from more than transmitter, with the signals side by side on the spectrum, is an interesting possibility.

MFSK16

Intuition might suggest that multiple-tone modes would not perform well when overlapped. But, as with some of the preceding results, MFSK16 continued to work surprisingly well and could tolerate the encroachment of other signals. For example, two MFSK16 signals were transmitted so that three-quarters of the audio spectrum was overlapping. Near perfect copy was received from both. However, this was almost definitely because the tones of the upper signal happened to be falling into the 15.625Hz gaps between the tones of the lower signal. This was proved by slightly shifting the upper signal in the spectrum, upon which copy was completely lost!

Reaction to *steady* tones was somewhat more interesting. So long as the unwanted tone frequency was outside the MFSK16 spectrum by more than 5Hz, no adverse effects were noticed.

When placed within the audio spectrum, it was found that single steady tones (such as stations tuning their transmitters) did cause interference until they were reduced in level by at least 3dB.

Intriguingly, a BPSK31 signal of equivalent signal strength was transmitted right in the middle of the MFSK16 spectrum [see **Figure 11**] where this caused absolutely no problem to perfect copy of the latter. This might have been because the BPSK31 waveform was not continuously present as was the case with a constant tone, and that the MFSK16 decoder could therefore still detect the tones it sought, in the gaps.

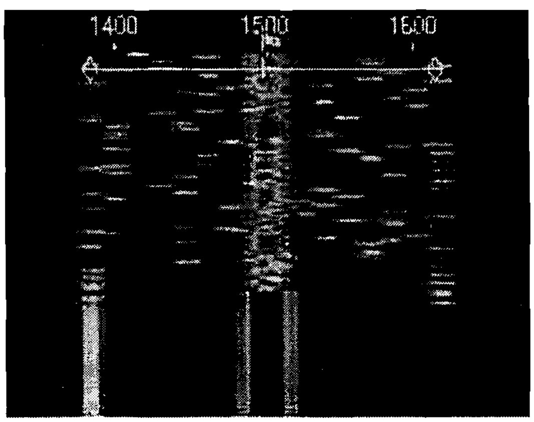

Figure 11: A strong BPSK31 transmission in the centre of the wanted MFSK16 signal

Remarkably, the MFSK16 signal was perfectly received. Out of interest, the BPSK31 signal was 75% correctly received, presumably because the MFSK16 tones were only occasional and fleeting across its much narrower spectrum.

MT63 modes

The severe overlapping of MT63 signals was effectively investigated in the delay tests above. Therefore, the challenge here (and it turned out to be quite so) was to prevent the MT63 from getting its message across.

Murray Greenman, in his excellent book Digital Modes For All Occasions [4], quite rightly describes MT63 as the 'juggernaut of data modes'! A variety of strong tones and other data signals were all generated alongside each other within the MT63 transmission. The resulting melange is shown in

Figure 12. Perhaps the reader is not surprised to learn that perfect copy was still extracted from the MT63 signal, although it took some considerable time before the text actually appeared at the receiver!

Perhaps it is also fair to say that the addition of just one more tone to the spectrum eventually muted all data reception.

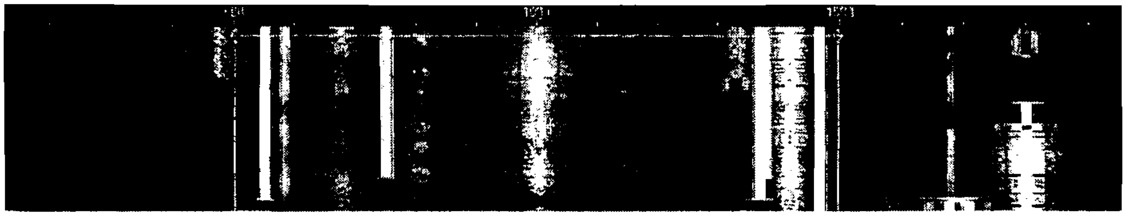

Figure 12. MT63 under extreme strain!

There are a total of six carrier tones and four PSK31 signals of various strengths within the pass-band, with a further healthy batch churning away nearby. Copy was still 100% after a twelve second number-crunching wait.

Interference from speech signals

All of the modes were tested here by mixing them with a sample of a heavily-modulated single sideband transmission [see **Figure 13**]. The waterfall display of the speech shows that, at any given moment, there are only small amounts of energy at any one spot frequency and that the waveform overall is transitory in nature. Therefore, none of the modes were all that bothered by the presence of the speech signal. The narrower the bandwidth of the mode the less it seemed to be affected.

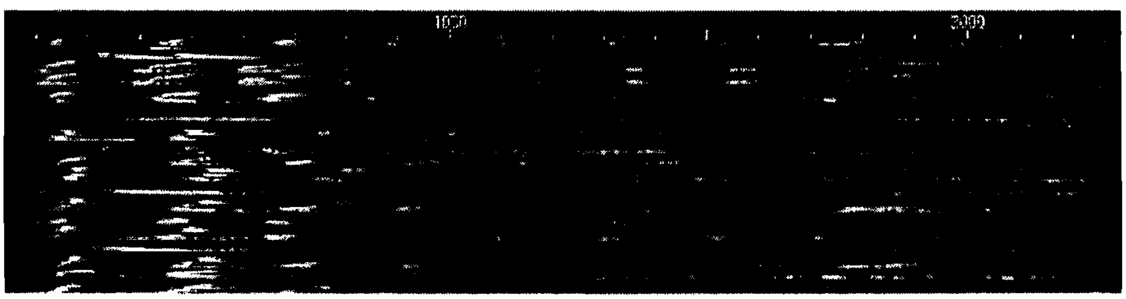

Figure 13. Single Sideband speech

The actual audio was a female voice, heavily compressed. Note that most of the energy is in the 500Hz region.

It is worth noting that although the data modes seemed fairly immune to speech, they were in turn very annoying to anyone who actually wished to listen to the speech content. MT63 was particularly wearing on the ear.

AGC considerations

It is important to be aware that all of the tests were conducted with no AGC in operation within the receiver. This is because the AGC acts across the whole audio spectrum and therefore a strong signal that is outside the bandwidth of the wanted data signal can nevertheless cause the receiver to severely desensitise. Although the decoding software is very tolerant of sudden changes in level, the effect is generally unhelpful when trying to receive weak signals.

Therefore, if possible, the AGC in the radio receiver should be completely switched out or at least reduced to the minimum setting. This can make a terrible noise in the receiver but will prevent out-of-band signals from clamping the receiver back by as much as 20dB.

If the facility is available, it is sometimes possible to reduce the receiver bandwidth through use of narrow IF or notch filters but their use should be approached with caution. In particular, DSP-based filtering can sometimes introduce unwanted by-products such as deep notches in the middle of the pass-band.

[4] RSGB Books ISBN 1 872309 82 8, the bible for the amateur data communicator and the boundless source of useful technical information for this investigation.

5. Efficiency (RF bandwidth and Duty Cycle)

Spectrum efficiency

Please refer to **Figure 2** and **Table 1** for the specification of the modes in question. As has been stated, whether the bandwidth used by the data mode is an issue is related to the channel available and the impact on any other users.

If it is envisaged that operations are to be conducted on normal amateur bands, then it may be necessary to compromise speed with the space taken up. Other users of the space are legitimately there and may feel less inclined towards emergency communications if there is some juicy 'DX' about. The PSK31 modes are most acceptable in this case, because, although the error correction is minimal, they use a small operating bandwidth where the chances of mutual interference are minimised.

If, however, en exclusive channel has been secured, the goal would be to use the permitted bandwidth to the absolute maximum. This would lead to higher data rates with much more comprehensive error correction systems that require a degree of redundancy to function. MT63 is the obvious mode of choice here, with the 2kHz version probably being the best choice unless conditions are very noisy.

Average transmitted power

Continuous broadcasting is not common in the amateur world. Therefore, compromises can be made in the transmission equipment that is of financial benefit to the amateur. These might include the downgrading of the power supply and the heatsink that dissipates heat away from the final output devices. So, while coping adequately with a longish SSTV transmission, the equipment may not survive, say, a seventy-two hour continuous broadcast. It should also be noted that overheating in older equipment could lead to frequency drifting, a serious problem in the field of data working.

An important factor to compensate for overheating problems is that data modes lend themselves to lower power levels than the equivalent speech requirement. Added to this is the fact that some (but by no means all) of the data modes are not of the constant amplitude type, meaning that there are significant portions of time when little radio frequency energy is being generated. Taking the average power output, therefore, is the illuminating measure of the efficiency of a data mode.

Table 3 gives a relative guide to average power level.

In determining power levels, it is important to measure this parameter accurately. Data modes are notorious for misleading most meters that are calibrated for speech ballistics. Refer to **Appendix A** for guidance on this matter.

As a rule, it has been found that an amateur rig will stay comfortably cool if the power output is kept to no more than 30% of the full available power. This might be increased to 50% in the case of modes that use a lower average power. There is a second good reason for doing this, which is that at these modest levels the operator can be sure that the ALC circuit in the transmitter is definitely not operating. This is very important to successful data transmission, and the procedure is outlined in **Appendix A**.

DATA MODE	AVERAGE POWER	COMMENTS
PSK31 modes	80%	Given the small bandwidth of these modes, power is concentrated within about 63Hz. The radio is likely to remain very cool! However, the temptation to increase power output must be resisted, as it is vital, but not easy, to maintain the greatest possible linearity. Unwanted extra sidebands alongside the PSK31 signal are commonly seen and imply distortion of the wanted signal.
MFSK16	100%	This mode produces full power output all of the time – thus there is no amplitude modulation. The transition between tones has no gap or phase change. This means that non-linear amplifiers can be employed and it also minimises the transmitted bandwidth. But the effect on the output stages of the radio is the same as holding down the Morse key constantly. As with long bouts of tuning tones, the amateur must be very careful not to burn out their equipment! It is therefore just as well that MFSK16 performs so well at low signal strengths.
MT63 modes	80%	Although the average power of these modes is the same as that of the PSK31 modes, it should be noted that this power is being developed over a wider bandwidth so the average signal strength of any one tone will be equivalently less. The bandwidth approximates to that used for speech but the presence of audio is more constant across that spectrum. Therefore MT63 places a higher demand on the equipment than speech.

Table 3: Average power output for the different modes

6. Ease of operation

At the risk of repetition, there is little advantage to proceeding with on-air assessment if the hardware set-up has been glossed over. Operating data modes generally requires a greater attention to detail in this area. Although correct setting of the transmit parameters is probably more critical than the receiving parameters, neither can be compromised. See **Appendix A** for more guidance in this area.

Table 4 gives a basic comparison. Refer to the text that follows for a more detailed discussion.

MODE	Sideband-sensitive?	Accuracy required for decoding	Ease of tuning on receive	Software set-up
BPSK31	No	Fairly easy	Easy	Easy
QPSK31	Yes	Fairly easy	Easy	Easy
MFSK16	Yes	Very critical	Difficult	Easy
MT63-1K	Yes	Not difficult	Fairly difficult	Fairly easy
MT63-2K	Yes	Not difficult	Fairly difficult	Fairly easy

Table 4: A comparison of the ease of operation

PSK31

Amongst the amateur fraternity, this group of modes is the most popular by far. BPSK31 predominates and it is often thought of as a 'starter' mode. This is probably because there are few parameters to set up and on-air use is generally easy as far as reception is concerned. There are many good versions of PSK software about and some of the newer variants, such as PSK125, are very fast while maintaining a friendly minimal bandwidth.

It is perhaps unfortunate, then, that arranging to transmit this mode requires fastidious attention to linearity throughout the audio and radio frequency chain.

BPSK31 is not sideband sensitive, which could be a useful benefit in ad hoc situations where there is limited knowledge at either end of the link.

Like most other data modes, QPSK31 *is* sideband sensitive. Tuning is very similar to its more popular counterpart.

The PSK31 signal is easy to spot, particularly when idling because it has two distinct 'tram lines'. The decoder cursor need only be placed fairly inaccurately within these carriers; the software AFC does the rest and copy should almost immediately result. Once data is being sent, the best attack for the receiving operator is to place the decoder cursor right in the middle of the fuzzy-looking band of the signal.

MFSK16

The undoubted benefits of MFSK16 are to some extent offset by the fact that it is really quite hard to tune in to. This is particularly difficult when the signal is weak, mainly because the individual tones are fleeting in nature and the highest and lowest tones can be very difficult to see in amongst the noise. It is very important that the transmitter sends healthy amounts of idle tone at the beginning of, and during, the transmission. The idle tone appears as a solid line on the lower edge of the signal, being the lowest tone in use.

As previously discussed, reception can be perfect even when the MFSK16 signal is well below the noise and barely visible on the waterfall display. Therefore, adjustments to the decoder frequency can be a 'suck it and see' affair, hampered further by the fact that correctly decoded text can take up to six seconds to start appearing on the screen! It is also the case that the operator must position the decoder cursor fairly closely to the correct audio frequency, otherwise the software AFC will not be able to home in on the waveform. Correct adjustment may be aided by zooming in on the waterfall display so that the MFSK16 spectrum is shown in as close a detail as possible.

Once reception has commenced, use of the software AFC is vital to keep the decoder locked. Even then, if deep fading is present, the decoder may lose the signal for long enough to start searching again which may require manual re-adjustment by the operator. The tuning dial of the radio itself should NOT be touched.

When transmitting AND receiving, RIT (receiver incremental tuning) should never be used, as an offset of more than 10Hz between the two stations of a contact will lead to a failure to decode at either end.

MFSK16 requires equipment that is very stable in frequency. This rules out the use of older transceivers that are often subject to this sort of magnitude of frequency drifting over time.

A possible benefit to MFSK16 is that a mechanical CW filter could be used to reduce unwanted signals and noise from outside the pass-band. However, as has been stated, it is generally unwise and not particularly helpful to filter signals at the radio frequency level.

MT63

Tuning of MT63 modes is not all that critical. This is because the mode can use Forward Error Correction techniques to examine different combinations of the 64 tones that calculate the correct location within the spectrum. As an example, MT63-1K will still work if the decoder is off tune by as much as 100Hz. MT63-2K is even less exacting, with an error of 250Hz being tolerated.

The main problem with MT63 modes, particularly at low signal strengths, is that the waveform can be very indistinct and sound just like noise. So this mode cannot be detected either by ear or eye. To get around this for marginal signal long distance work, convention dictates that MT63 signals will be transmitted on predetermined frequencies (see **Appendix C**) so that the position of the data stream in the audio spectrum can be assumed. For unprepared operation, the receiving operator can be deceived if there is any cancellation affecting only a part of the overall signal. Probably the best advice is to look carefully for the hard upper or lower edges of the signal.

Most MT63 programs do not offer the option to adjust the decoding frequency. Therefore, altering the radio receiver frequency may be necessary. However, once correct reception has started it is very unwise to adjust the radio frequency further, even by tiny amounts, as this can lead to a loss of synchronisation. Recovering from this state can take the software many seconds.

Another problem with the reception of MT63 is the fact that correctly decoded text can take up to 15 seconds to appear on the screen. A clue to synchronised reception can sometimes be gleaned if the software has a digital squelch facility. If the squelch is set up to be closed when no MT63 signal is present, it can often be seen to open as soon as it finds MT63 and ideally the text will follow quite a number of seconds later! Another clue to satisfactory reception prior to the actual appearance of the text is that the generation of large volumes of unwanted characters tends to cease when receiver synchronisation has been achieved.

As with other multi-tone modes where many sine waves exists side by side, the transmission chain for MT63 must be made as linear as possible.

7. Compatibility and adaptability

With all of the modes in question, the basic coding and decoding software systems are all compatible. Generally speaking, however, there is more than one choice of software for each mode. Compatibility issues can start to come to light if the sender and receiver choose to use different programs. The danger is that the sender may decide to use a facility that is only available with the program they are using, not also at the distant end. This might include the sending of binary files, or perhaps setting a specific parameter that is fixed at the other end of the link. Needless to say, all of these problems can be avoided by the use of the same software throughout the network. Even then, it is important to check version numbers as new facilities are being added with each issue.

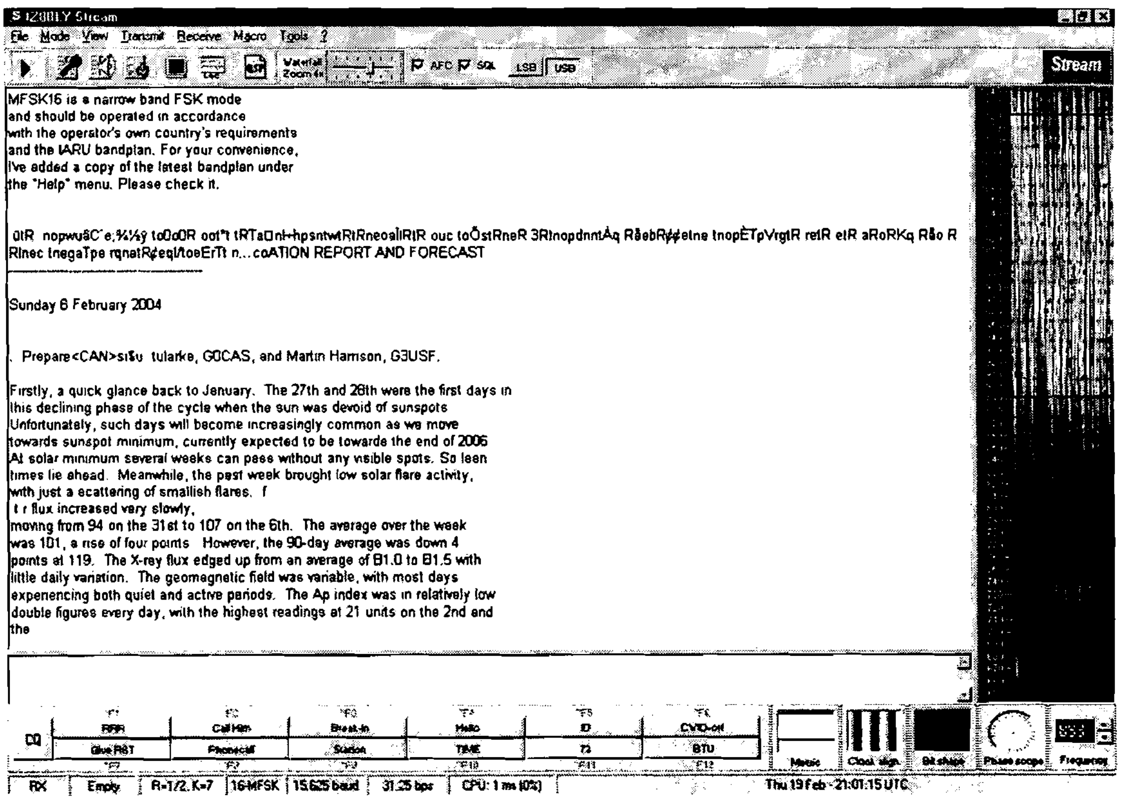

Figure 14: The IZ8BLY screen for MFSK16. Note the vertical waterfall display.

In terms of using the different modes to accurately relay documents or similar, it is worth briefly looking at the fact that they all use different character sets.

PSK31 has its own varicode [5] technique for sending the data, which uses its own unique character set.

MFSK16 uses an extended 8-bit X-ASCII character set (which also then translates the characters into its own varicode alphabet).

MT63 generally uses a 7-bit ITA-5 ASCII code. However, to expand the available alphabet (permitting extra control characters to be added) most emulations of the mode now allow the addition of an eighth data bit.

[5] The varicode concept is described as part of the DATA RATE results above.

8. Reliability

Much of the popular software has been run on a variety of computers of differing capabilities. Of the three modes, PSK31 makes the least demand on computer time and MT63 the most.

To run PSK31 successfully, a fairly modest computer is all that is required. For MFSK16, it is recommended that a Pentium processor would be the minimum specification. MT63-2K is a demanding mode and the processor should be running at 400MHz at least.

Importantly, these figures relate to the communications software *alone*. In reality, the computer is likely to be in use for many simultaneous tasks and therefore the processing time must be shared.

A good test is to generate an MT63 bitstream, then experiment with opening other software items on the same machine or undertake some cutting and pasting operations while listening to the tones. It has been noticed that, when the computer comes under time constraints, the tones can briefly disappear. Whether or not there is data actually being lost in these gaps is difficult to judge, but there is no doubt that the momentary loss of synchronisation leads to a noticeable loss of reception at the far end.

Unfortunately, cutting and pasting operations are likely to be a feature of an emergency communications station so work is needed to explore this area more thoroughly.

9. Availability

Via the Internet, there are several software packages, many of them freeware, which can be downloaded for use. For example, DIGIPAN and HAMSCOPE are popular choices. In particular, the suite of programs by IZ8BLY [6] and the excellent MIXW [7] are multi-mode and therefore offer many protocols to try within the one program.

[6] Nino Porcino, IZ8BLY, has produced a series of programs that concentrate on many specific modes. All of his software can be downloaded from his website http://iz8bly.sysonline.it/. The individual mode programs can inter-react. Nino has also produced the STREAM program which brings together many of the modes into one package.

[7] MIXW2 is a shareware program offering all of the popular data modes. The program can be downloaded from http://www.mixw.co.uk/ and can be used for a free trial period prior to registering for a modest fee.

Of all the modes, there are probably more software offerings for PSK31 than any other. About a dozen versions can be found on the Internet. For non-Windows users, there are Linux and Mac versions.

MFSK mode development is worth watching as there are continuing attempts to reduce the amount of Forward Error Correction in order to obtain higher speeds. There is one version of MFSK16 for Linux.

The two key MT63 programs are the stand-alone IZ8BLY program and as part of the MIXW package. These offer the basic data communication facility but differ in the enhancements offered. For example, the IZ8BLY version will allow the transfer of binary files and the use of the inherent secondary beacon channel but neither of these are offered by MIXW. Pawel Jalocha SP9VRC, the inventor of MT63, has produced a version for Linux.

As has been lamented already, none of these programs are particularly sophisticated when it comes to the handling of bulk text or binary files.

CONCLUSIONS AND RECOMMENDATIONS

Table of results

Table 5 is an attempt to represent the relative performance and attributes of the modes examined, hopefully in a simple form. Essentially, the various test results have been prioritised towards the specific requirements of emergency communications under different propagation conditions. Data rate and accuracy are always much more important in emergency communications than they might be for casual use on the amateur bands and are therefore given due precedence.

Under each criterion, a score has been allotted based on the test results as follows:

1 = Poor
2 = Fair
3 = Satisfactory
4 = Good
5 = Excellent

Different propagation conditions can be encountered at HF. Therefore, in order to achieve the best possible communication, for example, under weak signal conditions or on a congested channel, it would be advisable to choose the mode that has the most appropriate attributes to the situation.

Therefore, an overall score for each mode is not appropriate. Instead, the propagation and operating conditions should be assessed and the table then used to guide the user to the best mode for the job.

For clarity, the mode(s) scoring highest in each category have been highlighted in green.

CRITERION	BPSK31	QPSK31	MFSK16	MT63-1K	MT63-2K
Accuracy	2	3	4	5	5
Data rate	2	2	3	4	5
Weak signal performance	3	4	5	3	2
Tolerance to interference	2	2	4	5	5
Efficiency	3	3	1	3	3
Ease of operation	4	4	1	3	3
Compatibility/Adaptability	4	4	3	3	3
Reliability	5	5	4	3	3
Availability	5	4	4	3	3

Table 5: Relative performance table

In terms of importance to the field of emergency communications, the criteria are ordered from top to bottom with the most important factors at the top.

Recommendations

The key findings of this investigation are:

When conditions are good and audio bandwidth is not an issue, **MT63** (particularly the 2kHz version) using the 'long interleave' setting is the best mode to use. The data rate is very rapid and the multiple use of error correction techniques results in the most robust broadcast mode readily available to the amateur. Although it can be a little tricky to identify by ear or eye, the mode has a generous tolerance to tuning inaccuracies and its immunity to impulse noise is second to none.

When signals are weak or unstable, MT63 becomes difficult to discern from the noise and the wider pass-band leaves the mode susceptible to admitting too many unwanted audio products. Under these conditions, **MFSK16** hugely out-performs the other modes while still maintaining a respectable data rate. While only occupying a modest amount of spectrum, the error correction and time interleaving combine to recover the data from the most marginal signals, a full 10dB lower in the noise than MT63 can manage. One of the compromises with MFSK16 is the heavy burden of 100% power demand on the transmitter. It is also notoriously difficult to tune when barely visible on the waterfall display and very stable equipment is a prerequisite for successful operation.

When the channel is heavily congested with other traffic, MT63 and MFSK16 both cope well up to a point. However, if it is simply a matter of finding a small enough gap in the activity, then the minimal bandwidth of **BPSK31** and **QPSK31** are to be preferred. QPSK31 does offer better error correction but suffers with the need to detect four phase states rather than two. This often means that little improvement is noticed in practice. The PSK modes are very easy to tune in and work reliably on modest computers. Another great advantage of BPSK31 in particular is that it is the most well known and easily understood of the data modes. There are many sources of the software, too.

For the future

A 3kHz version of MT63 would be very interesting to assess. This would be a good bandwidth to choose for optimum use of the space available under current licensing which continues to think of a single speech channel allocation as being this wide.

In the broadcast application considered here, there would not be an issue if the arrival of the data were to be delayed by a few more seconds. Therefore, to improve accuracy without increasing bandwidth, it would be beneficial to add more time-based interleaving.

Not much exploration has yet taken place into some of the newer or more obscure variants of these data modes. For example, PSK125 is similar to PSK31 but by using two sidebands of 125Hz the data rate is dramatically increased. The greater bandwidth also allows the introduction of improved error correction. There is also PSK250, operating at a faster rate again. However, the error correction employed does not approach that of MT63 and, as with all PSK modes, success is limited by the need for stable propagation conditions and, as the data rate climbs, increasingly powerful computing facilities.

Charles Brain G4GUO is forging new paths into the adaptation of some military standards which are known to give astounding performance at HF, and it may well be that his PC-ALE software will lead to even more efficient use of the spectrum for radio amateurs.

To really make data mode communication viable in the eyes of the User Services, a great deal of work is needed to improve the interface between the coding/decoding software and the format of the material that they require to send over the amateur link. Bandwidth will always be an issue, so programmers also need to concentrate on how the *amount* of data that needs to be transmitted is kept to the minimum.

CREDITS

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APPENDIX A

A CHECKLIST TOWARDS ACHIEVING GOOD DATA COMMUNICATIONS

This is intended to be a brief guide to setting up a data station. It follows the signal path firstly from the computer through to the transmitter, then back from the receiver to the computer.

TRANSMISSION

Computer preparation

- **Only run programs that are necessary**, halt others.
- **Switch off normal computer sounds** (in Windows, select START, SETTINGS, CONTROL PANEL, SOUNDS, MULTIMEDIA then select 'No Sounds' in the 'Scheme' box).
- **Set up the Sound Card for transmission ('replay')** (in Windows, select START, PROGRAMS, ACCESSORIES, ENTERTAINMENT, VOLUME CONTROL then pull down the OPTIONS menu and select PROPERTIES, PLAYBACK). Mute all of the sound sources present except 'Wave Balance' and, of course, the 'Master Volume'. Set the faders for both of these to approximately half way. Pay attention to the stereo balance control too. Most soundcards accept a stereo input and the way in which the input connector to the computer is wired will affect the setting required here. Note: This panel can usually be reached directly from within the communications software.
- Some computers have a **hardware volume control** which is also in line with the audio output. This should be set to approximately half volume.

At this stage, it may be useful to start the data mode software for a quick pre-connection check. After selecting the mode desired, place the software into TRANSMIT. It should be possible to hear the waveform via the computer loudspeakers (or via the headphone jack if no speakers are present). Note: Within the software, it is likely that the output level for each mode is settable. Follow the program's menu system to locate this and check that it is selected to the default output volume (often a numeric value, typically '1.00').

- **Locate the computer audio output**. The audio output from the computer soundcard will probably emerge via the headphone output socket (or preferably, if there is one, via the 'line output' connector; or even better still, via a professional balanced line output for the very well endowed!). Note: If another program is running which is already using the soundcard, the data mode program may be denied access to it.

- **Locate the radio transmitter audio input**. The most obvious point of connection is into the microphone socket. However, many modern rigs have a high level input, usually on the rear panel, and this is definitely the one to use if possible.

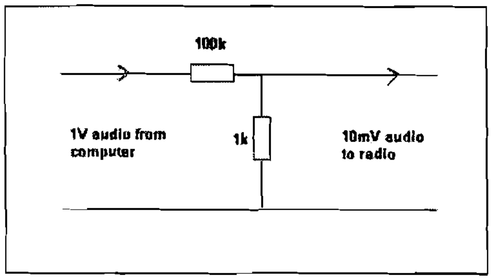
There are several reasons for this that are worthy of explanation in detail. The main issue is that the microphone input features a lot of gain (to raise the low microphone-level signals up to something more useful for the transmitter chain) and can be easily abused by the application of too much audio level. This input also expects to be connected via a short, heavily screened cable to the microphone itself. All amateurs are familiar with the problem of too much radio frequency energy within the shack – terrible feedback can occur and the point at which the unwanted RF enters the radio is almost certainly via the mic and cable, with the sensitivity of the microphone amplifier being a contributory factor.

A second reason for avoiding the microphone input is that a 'phantom' DC voltage is likely to be present. This is provided as a bias voltage for the common *electret* or condenser-type microphone capsules. Unless this voltage is DC-blocked, it will be applied directly to the computer audio output with unpredictable and potentially damaging results. Of course, primarily for electrical safety reasons, an audio transformer can be placed in the circuit to ensure no direct connection occurs. However, this is not entirely a good idea because, unless a great deal of money is spent on this item, the transformer will exhibit a limited audio bandwidth. It is also the case that the transformer can look just like a tuned inductive component to stray RF and can increase the likelihood of unwanted RF pick-up.

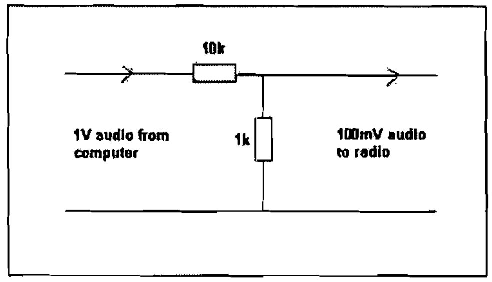
A third reason is that there will be a large amount of circuitry at an early point in the radio's microphone chain that is included to treat the incoming audio to make it more suitable for speech transmission. This may include audio compression, filtering and active (or even digital) speech processing. Whilst a boon for the speech communicator, these circuits can badly degrade a data signal because they were not designed with this form of audio in mind. It is likely that, although total avoidance of these circuits is unlikely, the high level input to the transmitter will enter the transmit chain at a much later point (ideally just before the modulation stages).

- **Determine the resistive network required**. Whatever the input and output connection points are, there will almost definitely be the need to provide audio level attenuation from the computer into the transmitter. It is fairly usual to use the headphone output of the computer, as this is often the only port available. This will produce a much higher level than even the high level input of the radio can cope with. It may be tempting simply to reduce the output volume from the computer, but this is unwise. Not only will the signal-to-noise ratio worsen undesirably, it is quite possible (depending on the quality of the soundcard) for this action to alter the spectral make-up of the transmitted audio. This is due to the impedance mismatch that the output is probably working into.

If good specifications of the computer output and radio input circuits are available, it is possible to work out the required attenuation circuit mathematically while satisfying the two port impedances. However, for most people this exercise is more likely to be a bit of 'suck it and see'. As a guide, some suggested starting values for a resistive network are given in **Figure 15** below:



Microphone level attenuator



High level attenuator

Figure 15: Suggested attenuator networks for driving the transmitter.

If too much level is still entering the radio, experiment by reducing the value of the 1k resistor.

As discussed briefly above, the most susceptible point to distortion in the transmitted audio chain is probably the first preamplifier after the microphone input. It is important not to overdrive this circuit. Although the turning down of the 'Mic Gain' control on the radio can always reduce audio levels, this is situated *after* the preamplifier circuit and is of no use if there is already too much level entering at the mic socket. A distorted signal is still just as distorted when it is reduced in volume and <u>once distortion has been introduced it cannot be removed by any method</u>.

When making up the cable linking the computer to the radio via the resistive network, good quality screened cable should be used to eliminate audio hum and the lengths should be kept to the absolute minimum. Wherever possible, the audio cable runs should be diverted away from any cables carrying the RF output signal. If RF induction is encountered, use of ferrite beads or rings may help but the only true answer is to get the RF out of the shack!

It should also be remembered that most computers provide an unbalanced stereo output, with a wire for each leg of the stereo plus a return 'ground' wire. If only one leg of the stereo is to be cabled up, remember to slide the stereo fader on the PLAYBACK mixer fully to the left or right to optimise the output.

One of the advantages of transmitting broadcast data modes is that the transmitter PTT (push-to-talk) line need not be controlled by the computer. This would be vital for ARQ modes. It is, of course, necessary to have a method of placing a transceiver into transmit mode when the microphone might be unplugged. It is likely, however, that this can be achieved with an alternative switch on the equipment or possibly via a rear-panel accessory contact closure. If none of these facilities exist, it will be necessary to wire up a separate shorting switch to the PTT line emerging at the microphone socket.

Transmitter preparation

- **Mute the microphone**. Of course, this will happen automatically if the microphone cable is removed to allow access to the audio inputs of the transmitter. If an alternative audio input is in use, it is important to mute the microphone as otherwise unwanted audio such as keyboard tapping, or speech, will be radiated along with the data signal. Muting the microphone is usually a selectable function of the equipment. If not, the microphone should be unplugged.
- **Switch compression OFF completely**. It is very important to remove any transmitter facilities that modify the audio. This also applies to any other forms of speech processing or filtering. Note: Some modern transceivers have the option to choose an analogue or digital path through

the audio stages of the transmitter. Whether one or other is more appropriate to data transmission is a subject for debate. Critical examination of the on-air waveform, while switching between the two options on an Icom transceiver, did not show conclusive differences.

- **Ensure that the transmitter ALC is NOT operating.** Automatic Level Control or 'ALC', being a means of level reduction designed to prevent the overdriving of the output stage of a transmitter, is a form of amplitude modulation in itself and therefore can degrade the data waveform. The point at which the ALC starts to operate is difficult to determine because its operation can be fleeting. Also, with data modes such as MT63, the audio level can gently rise and fall over a period of many seconds with the changing bitstream. Therefore, the advice is: never drive the transmitter to more than 30% of full power output.

  The best way to develop clean power in most modern transceivers is to have the RF power output control set to maximum, but to only apply enough audio input to modulate the transmitter to 30%. Note: This is definitely NOT the same process as setting the audio gain to maximum!

  An important point to remember is that even very modern transceivers may require slightly different levels of drive on each band in order to achieve the same power output. Therefore, a quick level test into a dummy load is always worthwhile when changing from one amateur band to another.

If any of the audio input gain controls, such as 'Mic Gain' or 'Drive', are close to either end of their travel then it may be necessary to make small changes to the audio output of the soundcard or even to consider a change to the resistive network.

- **Ensure that Upper Sideband has been selected!** This is the convention for all data modes on <u>all</u> HF bands.
- **Switch off any 'monitor' facility.** This is a feature of some transceivers whereby the audio applied to the transmitter can be peeled around and fed back through the final audio stages of the receiver to the headphones or loudspeaker during transmission. This allows the operator to listen to the outgoing audio. Of course, it can be useful to check the audio that is being sent, particularly if the data mode software can be operated in full duplex mode. However, it has been shown that doing this can lead to unpredictable reactions within the computer or radio that affect the audio chain, so this facility is therefore to be used with caution.

  If possible, monitor the transmitted audio on a separate receiver instead. Bear in mind that this receiver could be subject to overload due to its proximity to the transmitter, giving a false impression of the radiated quality. If possible, operate this receiver without an antenna and with any available RF attenuation switched in.

RECEPTION

Receiver preparation

- **Set the radio tuning dial to high accuracy.** Many transceivers with digital frequency displays allow the operator to adjust the resolution of the tuning dial. This determines how quickly the frequency changes when the dial is turned. For data operation, the dial should be set to the highest possible resolution so that even quite large physical movements only alter the frequency of operation very slightly.

- **Switch the receiver AGC off**, reduce it to a minimum or set it to a 'fast' decay. This action prevents signals from outside the data signal pass-band from randomly and heavily dipping the level of the wanted signal. Depending on the quality of the receiver in use, total removal of any AGC control may introduce undesirable distortion. In this case, use the smallest amount of AGC selectable.
- **Consider the use of RF, IF or AF filters**, but make sure that the wanted pass-band is not compromised by the introduction of unwanted modifications.
- **Do not use 'noise blanking' or 'noise reduction' systems**. The former attempts to remove impulse noise such as car ignitions; the latter is concerned to expand the signal-to-noise ratio in the receiver in an attempt to improve reception. Regretfully, both systems are liable to treat the wanted data signal as a noise source and will try to extract it from the audio to the detriment of data decoding.
- **Locate the receiver audio output**. It is much more useful if the receiver has a line output which is separate from the loudspeaker or headphone jacks. If this is the case, the operator does not lose any functionality and can monitor the receiver activity under all circumstances. Such an output is also likely to be at a fixed volume. Using the headphone output is the next best option although this will be at a much higher level, will be subject to the setting of the receiver volume control and will probably mute the receiver loudspeaker when plugged into.

Computer preparation

- **Locate the computer input**. The computer soundcard may have a variety of inputs. The one to choose, if available, is the 'line input'. This will be well matched to a receiver that has a 'line output' and can probably be connected directly. If the computer only has a microphone level input, a resistive attenuation network will be required between the receiver and the computer. This will also be the case if any of the higher level outputs from the radio are used.
- **Determine the resistive network required**. As with the situation on transmit, the values of the resistors could be calculated mathematically or arrived at by trial and error. There is less at stake on reception in that there is no danger of radiating a poor-quality signal. However, the need to maintain linearity is equally as important to successful data reception. Most data programs like to be driven quite hard on receive. Being too conservative with levels can lead to poor data decoding.
- **Set the soundcard for reception ('record')** (in Windows, select START, PROGRAMS, ACCESSORIES, ENTERTAINMENT, VOLUME CONTROL, then pull down the OPTIONS menu and select PROPERTIES, RECORDING). Only one sound source can be selected at a time. Select the appropriate source and fade up to approximately half way.

    It should also be remembered that most computers require an unbalanced stereo input, with a wire for each leg of the stereo plus a return 'ground' wire. If only one leg of the stereo is to be cabled up, remember to slide the stereo fader on the RECORD mixer fully to the left or right to optimise the input.

    If the microphone input is used, use the ADVANCED control to set the treble and bass faders to the middle.

As an interesting example of how robust the data modes can be, even when they are being used under the most disadvantageous circumstances, the following tale may amuse at least.

The author set a receiver in the shack to an MT63 signal. A cheap radio frequency baby alarm picked up the receiver's loudspeaker audio on its internal microphone and transmitted it, just as it would a crying baby. At the other end of the house, the baby alarm receiver reproduced the audio on its internal loudspeaker. Next to this was loosely dangled a domestic quality microphone that was plugged into a computer running the MIXW program set to receive MT63. Considering the number of audio translations that the signal had passed through, let alone considering the very mediocre quality of the components in the chain, it was amazing to see perfect copy arriving on the distant computer. This was even more impressive because people were moving around in both rooms, causing clunks and speech to be passed over the 'link' as well as the wanted signal. What a credit to Forward Error Correction!

APPENDIX B

COOLEDIT PRO Version 2

CoolEdit is a professional audio program from the Syntrillium Software Corporation. It is used by sound engineers and broadcasters to edit, manipulate and mix sound. It records WAV files at very high bit rates and is therefore very useful as a test facility because the starting material is known to be of good quality. The program also features many useful effects, including variable delays, filters, pitch adjustment and compression. All of these adjustments can be made with scientific accuracy.

The program has an Edit Window, where sound samples can be treated individually as required. Then, sound samples can be mixed in the Multitrack Window and their relative levels and time coincidence can be adjusted at will. In this way, for example, signals can be mixed with noise, subjected to delay and filtering, all at the same time or individually.

Figure 16 shows a typical multitrack window in CoolEdit.

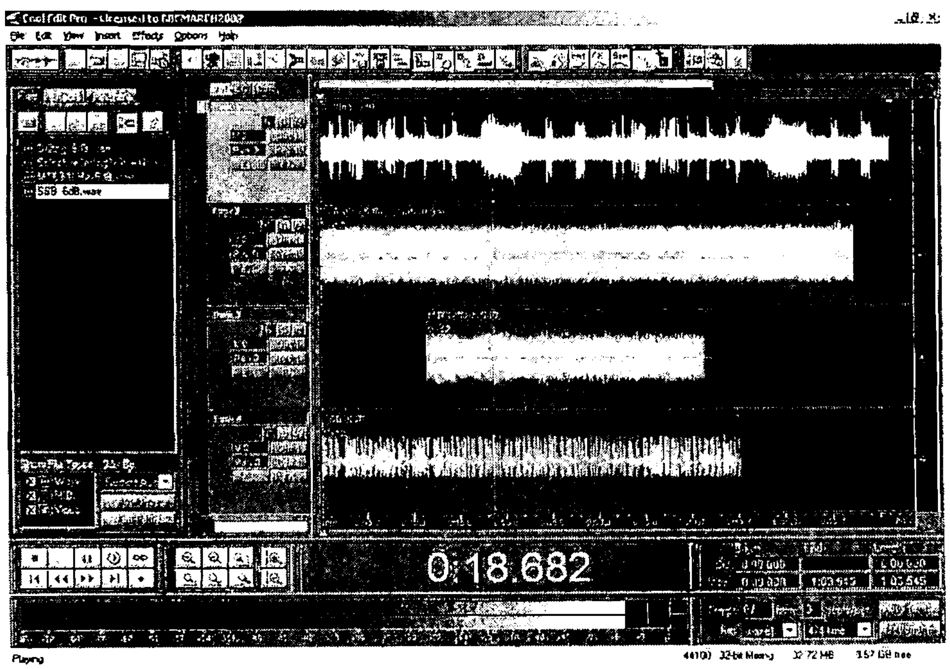


Figure 16: The Multitrack window of CoolEdit Pro v2

The impressive array of audio treatments can be seen lined up along the top of the screen. Much of the broadcasting output heard today, particularly in the fields of News and Sport, will have been prepared using this package.

The various waveforms can be seen as separate tracks; the cursor then moves across the screen in time as they are all replayed together. The resulting sound can be saved as a complete sound file, or it can be replayed through the computer soundcard into another machine on which data decoding software is running.

The program can record any amount of audio material, up to the limit of the hard drive capacity. The approximate memory required for CD quality stereo music is approximately 1GB per hour.

More information about this software and other audio products is available on the manufacturer's website at www.syntrillium.com.

APPENDIX C

DATA MODE FREQUENCIES GENERALLY USED ON AMATEUR HF BANDS

Table 6 shows the frequencies at which amateur data mode operations take place. This list was proposed internationally and appears to have been taken up by most amateurs, although it has yet to be fully ratified on a global scale. These frequencies are the places to look when seeking to receive a specific mode.

Remember that the data enthusiasts will be communicating quite happily on bands that would be described as 'closed' for speech modes. It is also worth remembering that some of the more exotic modes are by no means present all of the time.

The main centres of activity seem to be 20m and 80m (the latter more so after dark). The spot frequencies given are where to centre the receiver dial. The waterfall display on the computer screen will probably show several transmissions, side by side. The software decoder can then be tuned on the computer screen to the signal of interest. Note that it is not good practice to tune the radio receiver itself and it is particularly bad practice to switch in any RIT (receiver incremental tuning) if you also intend to transmit. It should be noted that when using some of the wider bandwidth modes, for example MT63, the software in use might fix the decoding window to a particular segment, in this case 500Hz-1500Hz. If this is the situation, it becomes permissible to tune the radio carefully to the desired signal.

The absolute beginner might like to start with BPSK31 on 14070kHz. There is nearly always activity on and around this frequency. Remember that the convention is that Upper Sideband is invariably used, even on the lower HF bands where speech reverts traditionally to LSB! If the radio forces the use of LSB on the lower bands, all is not lost because the software usually has an 'Invert' function which can be selected to invert the bitstream instead of the transmitter sideband.

MODE	160m	80m	40m	30m	20m	17m	15m	12m	10m
RTTY/PACTOR	1800	3580	7040	10130	14070	18100	21070	24920	28070
	1840	3620	7100	10140	14100	18105	21100	24925	28150
PSK31	1838	3580	7035	10140	14070	18100	21080	24920	28070
			7070						28120
MFSK/THROB	1838	3580	7037	10147	14080	18105	21080	24929	28080
MT63	1822	3580.15	7035.15	10140.15	14106.3	18105	21130	24925	28130
	1838.15	3590	7037	10145	14109.3	18100.15			
		3635			14114				
HELLSCHREIBER		3575	7030	10135	14063	18100	21070		28063
		3580	7035	10137	14064	18105			28070
		3559	7037	10145	14070				28100
			7040						28110
									28120
SSTV		3730			14227		21340		28675
					14230		21335		28680
FAX		3730			14227		21335		28675
PACKET		3620	7068.9	10124.9	14062.9	18101.9	21072.9	24925	
		3620.9	7070.9	10125.9	14072.9	18105	21100	24930	
		3623.9	7071.9	10126.9	14073.9	18107.9	21110		
		3627.9	7072.9	10133.9	14074.9	18110			
		3635	7073.9	10136.9	14075.9				
		3638.9	7076.9	10140	14076.9				
				10150	14095				
					14099.5				

Table 6: Common frequencies for amateur data communications (with thanks to AD4JE)