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The Transmission of IP Over the Vertical Blanking Interval of a Television Signal

Status of this Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

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1. Abstract

This document describes a method for broadcasting IP data in a unidirectional manner using the vertical blanking interval of television signals. It includes a description for compressing IP headers on unidirectional networks, a framing protocol identical to SLIP, a forward error correction scheme, and the NABTS byte structures.

2. Introduction

This RFC proposes several protocols to be used in the transmission of IP datagrams using the Vertical Blanking Interval (VBI) of a television signal. The VBI is a non-viewable portion of the television signal that can be used to provide point-to-multipoint IP data services which will relieve congestion and traffic in the traditional Internet access networks. Wherever possible these protocols make use of existing RFC standards and non-standards.

Traditionally, point-to-point connections (TCP/IP) have been used even for the transmission of broadcast type data. Distribution of the same content--news feeds, stock quotes, newsgroups, weather

reports, and the like--are typically sent repeatedly to individual clients rather than being broadcast to the large number of users who want to receive such data.

Today, IP is quickly becoming the preferred method of distributing one-to-many data on intranets and the Internet. The coming availability of low cost PC hardware for receiving television signals accompanied by broadcast data streams makes a defined standard for the transmission of data over traditional broadcast networks imperative. A lack of standards in this area as well as the expense of hardware has prevented traditional broadcast networks from becoming effective deliverers of data to the home and office.

This document describes the transmission of IP using the North American Basic Teletext Standard (NABTS), a recognized and industry-supported method of transporting data on the VBI. NABTS is traditionally used on 525-line television systems such as NTSC. Another byte structure, WST, is traditionally used on 625-line systems such as PAL and SECAM. These generalizations have exceptions, and countries should be treated on an individual basis. These existing television system standards will enable the television and Internet communities to provide inexpensive broadcast data services. A set of existing protocols will be layered above the specific FEC for NABTS including a serial stream framing protocol similar to SLIP (RFC 1055 [Romkey 1988]) and a compression technique for unidirectional UDP/IP headers.

The protocols described in this document are intended for the unidirectional delivery of IP datagrams using the VBI. That is, no return channel is described, or for that matter possible, in the VBI.

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119.

3. Proposed protocol stack

The following protocol stack demonstrates the layers used in the transmission of VBI data. Each layer has no knowledge of the data it encapsulates, and is therefore abstracted from the other layers. At the link layer, the NABTS protocol defines the modulation scheme used to transport data on the VBI. At the network layer, IP handles the movement of data to the appropriate clients. In the transport layer, UDP determines the flow of data to the appropriate processes and applications.

3.1. VBI

The characteristics and definition of the VBI is dependent on the television system in use, be it NTSC, PAL, SECAM or some other. For more information on Television standards worldwide, see ref [12].

3.1.1. 525 line systems

A 525-line television frame is comprised of two fields of 262.5 horizontal scan lines each. The first 21 lines of each field are not part of the visible picture and are collectively called the Vertical Blanking Interval (VBI).

Of these 21 lines, the first 9 are used while repositioning the cathode ray to the top of the screen, but the remaining lines are available for data transport.

There are 12 possible VBI lines being broadcast 60 times a second (each field 30 times a second). In some countries Line 21 is reserved for the transport of closed captioning data (Ref.[11]). In that case, there are 11 possible VBI lines, some or all of which could be used for IP transport. It should be noted that some of these lines are sometimes used for existing, proprietary, data and testing services. IP delivery therefore becomes one more data service using a subset or all of these lines.

3.1.2. 625 Line Systems

A 625-line television frame is comprised of two fields of 312.5 horizontal scan lines each. The first few lines of each field are used while repositioning the cathode ray to the top of the screen. The lines available for data insertion are 6-22 in the first field and 319-335 in the second field.

There are, therefore, 17 possible VBI lines being broadcast 50 times a second (each field 25 times a second), some or all of which could be used for IP transport. It should be noted that some of these lines are sometimes used for existing, proprietary, data and testing services. IP, therefore, becomes one more data service using a subset or all of these lines.

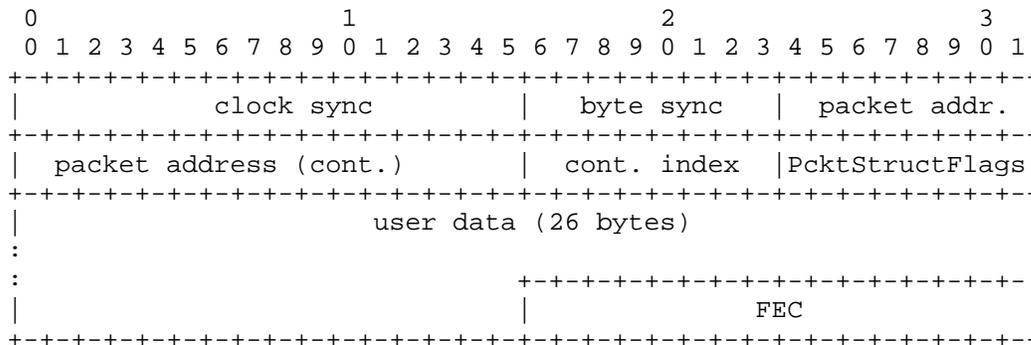
3.2. NABTS

The North American Basic Teletext Standard is defined in the Electronic Industry Association's EIA-516, Ref. [2], and ITU.R BT.653-2, system C, Ref. [13]. It provides an industry-accepted

method of modulating data onto the VBI, usually of an NTSC signal. This section describes the NABTS packet format as it is used for the transport of IP.

It should be noted that only a subset of the NABTS standard is used, as is common practice in NABTS implementations. Further information concerning the NABTS standard and its implementation can be found in EIA-516.

The NABTS packet is a 36-byte structure encoded onto one horizontal scan line of a television signal having the following structure:



The two-byte Clock Synchronization and one-byte Byte Synchronization are located at the beginning of every scan line containing a NABTS packet and are used to synchronize the decoding sampling rate and byte timing.

The three-byte Packet Address field is Hamming encoded (as specified in EIA-516), provides 4 data bits per byte, and thus provides 4096 possible packet addresses. These addresses are used to distinguish related services originating from the same source. This is necessary for the receiver to determine which packets are related, and part of the same service. NABTS packet addresses therefore distinguish different data services, possibly inserted at different points of the transmission system, and most likely totally unrelated. Section 4 of this document discusses Packet Addresses in detail.

The one-byte Continuity Index field is a Hamming encoded byte, which is incremented by one for each subsequent packet of a given Packet Address. The value or number of the Continuity Index sequences from 0 to 15. It increments by one each time a data packet is transmitted. This allows the decoder to determine if packets were lost during transmission.

The Packet Structure field is also a Hamming encoded byte, which contains information about the structure of the remaining portions of the packet. The least significant bit is always "0" in this implementation. The second least significant bit specifies if the Data Block is full--"0" indicates the data block is full of useful data, and "1" indicates some or all of the data is filler data. The two most significant bits are used to indicate the length of the suffix of the Data Block--in this implementation, either 2 or 28 bytes (10 for 2-byte FEC suffix, 11 for 28-byte FEC suffix). This suffix is used for the forward error correction described in the next section. The following table shows the possible values of the Packet Structure field:

Data Packet, no filler	D0
Data Packet, with filler	8C
FEC Packet	A1

The Data Block field is 26 bytes, zero to 26 of which is useful data (part of a IP packet or SLIP frame), the remainder is filler data. Data is byte-ordered least significant bit first. Filler data is indicated by an 0x15 followed by as many 0xEA as are needed to fill the Data Block field. Sequential data blocks minus the filler data form an asynchronous serial stream of data.

These NABTS packets are modulated onto the television signal sequentially and on any combination of lines.

3.3. FEC

Due to the unidirectional nature of VBI data transport, Forward Error Correction (FEC) is needed to ensure the integrity of data at the receiver. The type of FEC described here and in the appendix of this document for NABTS has been in use for a number of years, and has proven popular with the broadcast industry. It is capable of correcting single-byte errors and single- and double-byte erasures in the data block and suffix of a NABTS packet. In a system using NABTS, the FEC algorithm splits a serial stream of data into 364-byte "bundles". The data is arranged as 14 packets of 26 bytes each. A function is applied to the 26 bytes of each packet to determine the two suffix bytes, which (with the addition of a header) complete the NABTS packet (8+26+2).

For every 14 packets in the bundle, two additional packets are appended which contain only FEC data, each of which contain 28 bytes of FEC data. The first packet in the bundle has a Continuity Index value of 0, and the two FEC only packets at the end have Continuity Index values of 14 and 15 respectively. This data is obtained by first writing the packets into a table of 16 rows and 28 columns.

In the case where the most significant bit in this field is set to "1", the length of the field extends to two bytes, allowing for 32768 additional schemas:

extended schema 2 bytes	packet (formatted according to schema) ?? bytes (schema dependant length)
-------------------------------	--

In the section 3.5, one such schema for sending IP is described. This is the only schema specified by this document. Additional schemas may be proposed for other packet types or other compression schemes as described in section 7.

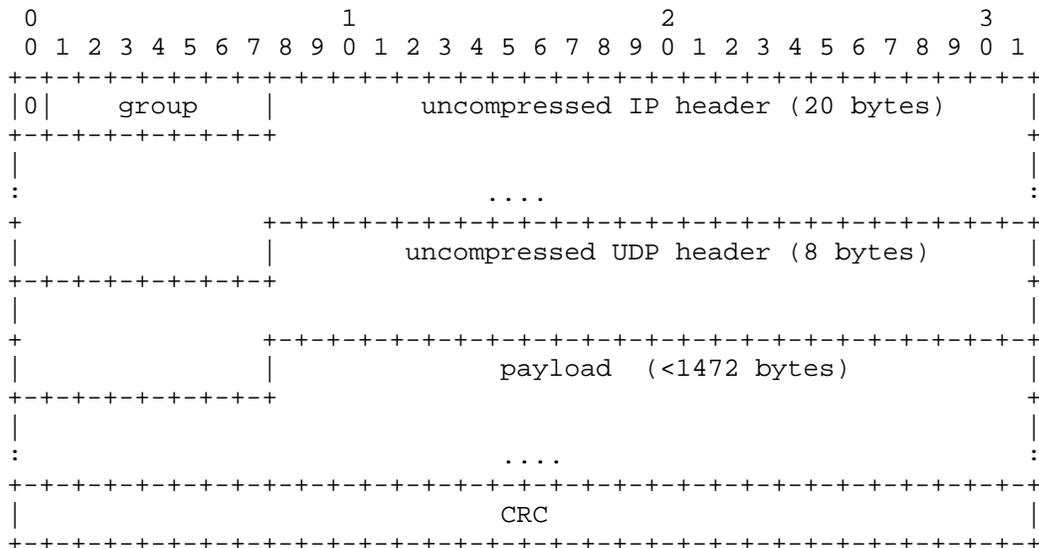
3.4.1 Maximum Transmission Unit Size

The maximum length of an uncompressed IP packet, or Maximum-Transmission Unit (MTU) size is 1500 octets. Packets larger than 1500 octets MUST be fragmented before transmission, and the client VBI interface MUST be able to receive full 1500 octet packet transmissions.

3.5. IP Header Compression Scheme

The one-byte scheme defines the format for encoding the IP packet itself. Due to the value of bandwidth, it may be desirable to introduce as much efficiency as possible in this encoding. One such efficiency is the optional compression of the UDP/IP header using a method related to the TCP/IP header compression as described by Van Jacobson (RFC 1144). UDP/IP header compression is not identical due to the limitation of unidirectional transmission. One such scheme is proposed in this document for the compression of UDP/IP version 4. It is assigned a value of 0x00. All future encapsulation schemes should use a unique value in this field.

Only schema 0x00 is defined in this document; this schema must be supported by all receivers. In schema 0x00, the format of the IP packet itself takes one of two forms. Packets can be sent with full, uncompressed headers as follows:



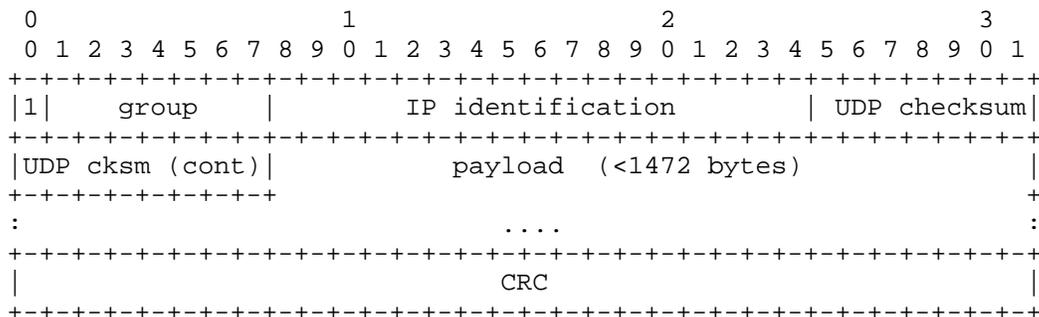
The first byte in the 0x00 scheme is the Compression Key. It is a one-byte value: the first bit indicates if the header has been compressed, and the remaining seven bits indicate the compression group to which it belongs.

If the high bit of the Compression Key is set to zero, no compression is performed and the full header is sent, as shown above. The client VBI interface should store the most recent uncompressed header for a given group value for future potential use in rebuilding subsequent compressed headers. Packets with identical group bits are assumed to have identical UDP/IP headers for all UDP and IP fields, with the exception of the "IP identification" and "UDP checksum" fields.

Group values may be recycled following 60 seconds of nonuse by the preceding UDP/IP session (no uncompressed packets sent), or by sending packets with uncompressed headers for the 60-second duration following the last packet in the preceding UDP/IP session. Furthermore, the first packet sent following 60 seconds of nonuse, or compressed header packets only use, must use an uncompressed header. Client VBI interfaces should disregard compressed packets received 60 or more seconds after the last uncompressed packet using a given group address. This avoids any incorrectly decompressed packets due to group number reuse, and limits the outage due to a lost uncompressed packet to 60 seconds.

If the high bit of the Compression Key is set to one, a compressed version of the UDP/IP header is sent. The client VBI interface must then combine the compressed header with the stored uncompressed

header of the same group and recreate a full packet. For compressed packets, the only portions of the UDP/IP header sent are the "IP identification" and "UDP checksum" fields:



All datagrams belonging to a multi fragment IP packet shall be sent with full headers, in the uncompressed header format. Therefore, only packets that have not been fragmented can be sent with the most significant bit of the compression key set to "1".

A 32-bit CRC has also been added to the end of every packet in this scheme to ensure data integrity. It is performed over the entire packet including schema byte, compression key, and either compressed or uncompressed headers. It uses the same algorithm as the MPEG-2 transport stream (ISO/IEC 13818-1). The generator polynomial is:

$$1 + D + D^2 + D^4 + D^5 + D^7 + D^8 + D^{10} + D^{11} + D^{12} + D^{16} + D^{22} + D^{23} + D^{26} + D^{32}$$

As in the ISO/IEC 13818-1 specification, the initial state of the sum is 0xFFFFFFFF. This is not the same algorithm used by Ethernet. This CRC provides a final check for damaged datagrams that span FEC bundles or were not properly corrected by FEC.

4. Addressing Considerations

The addressing of IP packets should adhere to existing standards in this area. The inclusion of an appropriate source address is needed to ensure the receiving client can distinguish between sources and thus services if multiple hosts are sharing an insertion point and NABTS packet address.

NABTS packet addressing is not regulated or standardized and requires care to ensure that unrelated services on the same channel are not broadcasting with the same packet address. This could occur due to multiple possible NABTS insertion sites, including show production, network redistribution, regional operator, and local operator.

Traditionally, the marketplace has recognized this concern and made amicable arrangements for the distribution of these addresses for each channel.

5. IANA Considerations

IANA will register new schemas on a "First Come First Served" basis [RFC 2434]. Anyone can register a scheme, so long as they provide a point of contact and a brief description. The scheme number will be assigned by IANA. Registrants are encouraged to publish complete specifications for new schemas (preferably as standards-track RFCs), but this is not required.

6. Security Considerations

As with any broadcast network, there are security issues due to the accessibility of data. It is assumed that the responsibility for securing data lies in other protocol layers, including the IP Security (IPSEC) protocol suite, Transport Layer Security (TLS) protocols, as well as application layer protocols appropriate for a broadcast-only network.

7. Conclusions

This document provides a method for broadcasting Internet data over a television signal for reception by client devices. With an appropriate broadcast content model, this will become an attractive method of providing data services to end users. By using existing standards and a layered protocol approach, this document has also provided a model for data transmission on unidirectional and broadcast networks.

8. Acknowledgements

The description of the FEC in Appendix A is taken from a document prepared by Trevor Dee of Norpak Corporation. Dean Blackketter of WebTV Networks, Inc., edited the final draft of this document.

9. References

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10. Acronyms

FEC	- Forward Error Correction
IP	- Internet Protocol
NABTS	- North American Basic Teletext Standard
NTSC	- National Television Standards Committee
NTSC-J	- NTSC-Japanese
PAL	- Phase Alternation Line
RFC	- Request for Comments
SECAM	- Sequentiel Couleur Avec Memoire (sequential color with memory)
SLIP	- Serial Line Internet Protocol
TCP	- Transmission Control Protocol
UDP	- User Datagram Protocol
VBI	- Vertical Blanking Interval
WST	- World System Teletext

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12. Appendix A: Forward Error Correction Specification

This FEC is optimized for data carried in the VBI of a television signal. Teletext has been in use for many years and data transmission errors have been categorized into three main types: 1) Randomly distributed single bit errors 2) Loss of lines of NABTS data 3) Burst Errors

The quantity and distribution of these errors is highly variable and dependent on many factors. The FEC is designed to fix all these types of errors.

12.1. Mathematics used in the FEC

Galois fields form the basis for the FEC algorithm presented here. Rather than explain these fields in general, a specific explanation is given of the Galois field used in the FEC algorithm.

The Galois field used is $GF(2^8)$ with a primitive element alpha of 00011101. This is a set of 256 elements, along with the operations of "addition", "subtraction", "division", and "multiplication" on these elements. An 8-bit binary number represents each element.

The operations of "addition" and "subtraction" are the same for this Galois field. Both operations are the XOR of two elements. Thus, the "sum" of 00011011 and 00000111 is 00011100.

Division of two elements is done using long division with subtraction operations replaced by XOR. For multiplication, standard long multiplication is used but with the final addition stage replaced with XOR.

All arithmetic in the following FEC is done modulo 100011101; for instance, after you multiply two numbers, you replace the result with its remainder when divided by 100011101. There are 256 values in this field (256 possible remainders), the 8-bit numbers. It is very important to remember that when we write $A*B = C$, we more accurately imply $\text{modulo}(A*B) = C$.

It is obvious from the above description that multiplication and division is time consuming to perform. Elements of the Galois Field share two important properties with real numbers.

$$A*B = \text{POWERalpha}(\text{LOGalpha}(A) + \text{LOGalpha}(B))$$

$$A/B = \text{POWERalpha}(\text{LOGalpha}(A) - \text{LOGalpha}(B))$$

The Galois Field is limited to 256 entries so the power and log tables are limited to 256 entries. The addition and subtraction shown is standard so the result must be modulo alpha. Written as a "C" expression:

```
A*B = apower[alog[A] + alog[B]]
A/B = apower[255 + alog[A] - alog[B]]
```

You may note that $\text{alog}[A] + \text{alog}[B]$ can be greater than 255 and therefore a modulo operation should be performed. This is not necessary if the power table is extended to 512 entries by repeating the table. The same logic is true for division as shown. The power and log tables are calculated once using the long multiplication shown above.

12.2. Calculating FEC bytes

The FEC algorithm splits a serial stream of data into "bundles". These are arranged as 16 packets of 28 bytes when the correction bytes are included. The bundle therefore has 16 horizontal codewords interleaved with 28 vertical codewords. Two sums are calculated for a codeword, S0 and S1. S0 is the sum of all bytes of the codeword each multiplied by alpha to the power of its index in the codeword. S1 is the sum of all bytes of the codeword each multiplied by alpha to the power of three times its index in the codeword. In "C" the sum calculations would look like:

```
Sum0 = 0;
Sum1 = 0;
For (i = 0; i < m; i++)
{
    if (codeword[i] != 0)
    {
        Sum0 = sum0 ^ power[i + alog[codeword[i]]];
        Sum1 = sum1 ^ power[3*i + alog[codeword[i]]];
    }
}
```

Note that the codeword order is different from the packet order. Codeword positions 0 and 1 are the suffix bytes at the end of a packet horizontally or at the end of a column vertically.

When calculating the two FEC bytes, the summation above must produce two sums of zero. All codewords except 0 and 1 are known so the sums for the known codewords can be calculated. Let's call these values tot0 and tot1.

```
Sum0 = tot0^power[0+alog[codeword[0]]]^power[1+alog[codeword[1]]]
Sum1 = tot1^power[0+alog[codeword[0]]]^power[3+alog[codeword[1]]]
```

This gives us two equations with the two unknowns that we can solve:

```
codeword[1] = power[255+alog[tot0^tot1]-alog[power[1]^power[3]]]
codeword[0] = tot0^power[alog[codeword[1]]+alog[power[1]]]
```

12.3. Correcting Errors

This section describes the procedure for detecting and correcting errors using the FEC data calculated above. Upon reception, we begin by rebuilding the bundle. This is perhaps the most important part of the procedure because if the bundle is not built correctly it cannot possibly correct any errors. The continuity index is used to determine the order of the packets and if any packets are missing (not captured by the hardware). The recommendation, when building the bundle, is to convert the bundle from packet order to codeword order. This conversion will simplify the codeword calculations. This is done by taking the last byte of a packet and making it the second byte of the codeword and taking the second last byte of a packet and making it the first byte of a codeword. Also the packet with continuity index 15 becomes codeword position one and the packet with continuity index 14 becomes codeword position zero. The same FEC is used regardless of the number of bytes in the codeword. So let's think of the codewords as an array `codeword[vert][hor]` where `vert` is 16 packets and `hor` is 28. Each byte in the array is protected by both a horizontal and a vertical codeword. For each of the codewords, the sums must be calculated. If both the sums for a codeword are zero then no errors have been detected for that codeword. Otherwise, an error has been detected and further steps need to be taken to see if the error can be corrected. In "C" the horizontal summation would look like:

```

for (i = 0; i < 16; i++)
{
    sum0 = 0;
    sum1 = 0;
    for (j = 0; j < hor; j++)
    {
        if (codeword[i][j] != 0)
        {
            sum0 = sum0 ^ power[j + alog[codeword[i][j]]];
            sum1 = sum1 ^ power[3*j + alog[codeword[i][j]]];
        }
    }
    if ((sum0 != 0) || (sum1 != 0))
    {
        Try Correcting Packet
    }
}

```

Similarly, vertical looks like:

```

for (j = 0; i < hor; i++)
{
    sum0 = 0;
    sum1 = 0;
    for (i = 0; i < 16; i++)
    {
        if (codeword[i][j] != 0)
        {
            sum0 = sum0 ^ power[i + alog[codeword[i][j]]];
            sum1 = sum1 ^ power[3*i + alog[codeword[i][j]]];
        }
    }
    if ((sum0 != 0) || (sum1 != 0))
    {
        Try Correcting Column
    }
}

```

12.4. Correction Schemes

This FEC provides four possible corrections:

- 1) Single bit correction in codeword. All single bit errors.
- 2) Double bit correction in a codeword. Most two-bit errors.
- 3) Single byte correction in a codeword. All single-byte errors.
- 4) Packet replacement. One or two missing packets from a bundle.

12.4.1. Single Bit Correction

When correcting a single-bit in a codeword, the byte and bit position must be calculated. The equations are:

$$\text{Byte} = 1/2\text{LOG}\alpha(S1/S0)$$

$$\text{Bit} = 8\text{LOG}\alpha(S0/\text{POWER}\alpha(\text{Byte}))$$

In "C" this is written:

```
Byte = alog[power[255 + alog[sum1] - alog[sum0]]];
if (Byte & 1)
    Byte = Byte + 255;
Byte = Byte >> 1;
Bit = alog[power[255 + alog[sum0] - Byte]] << 3;
while (Bit > 255)
    Bit = Bit - 255;
```

The error is correctable if Byte is less than the number of bytes in the codeword and Bit is less than eight. For this math to be valid both sum0 and sum1 must be non-zero. The codeword is corrected by:

```
codeword[Byte] = codeword[Byte] ^ (1 << Bit);
```

12.4.2. Double Bit Correction

Double bit correction is much more complex than single bit correction for two reasons. First, not all double bit errors are deterministic. That is two different bit patterns can generate the same sums. Second, the solution is iterative. To find two bit errors you assume one bit in error and then solve for the second error as a single bit error.

The procedure is to iteratively move through the bits of the codeword changing each bit's state. The new sums are calculated for the modified codeword. Then the single bit calculation above determines if this is the correct solution. If not, the bit is restored and the next bit is tried.

For a long codeword, this can involve many calculations. However, tricks can speed the process. For example, the vertical sums give a strong indication of which bytes are in error horizontally. Bits in other bytes need not be tried.

12.4.3. Single Byte Correction

For single byte correction, the byte position and bits to correct must be calculated. The equations are:

$$\text{Byte} = 1/2 * \text{LOGalpha}(S1/S0)$$

$$\text{Mask} = S0 / \text{POWERalpha}[\text{Byte}]$$

Notice that the byte position is the same calculation as for single bit correction. The mask will allow more than one bit in the byte to be corrected. In "C" the mask calculation looks like:

$$\text{Mask} = \text{power}[255 + \text{alog}[\text{sum0}] - \text{Byte}]$$

Both sum0 and sum1 must be non-zero for the calculations to be valid. The Byte value must be less than the codeword length but Mask can be any value. This corrects the byte in the codeword by:

$$\text{Codeword}[\text{Byte}] = \text{Codeword}[\text{Byte}] \wedge \text{Mask}$$

12.4.4. Packet Replacement

If a packet is missing, as determined by the continuity index, then its byte position is known and does not need to be calculated. The formula for single packet replacement is therefore the same as for the Mask calculation of single byte correction. Instead of XORing an existing byte with the Mask, the Mask replaces the missing codeword position:

$$\text{Codeword}[\text{Byte}] = \text{Mask}$$

When two packets are missing, both the codeword positions are known by the continuity index. This again gives two equations with two unknowns, which is solved to give the following equations. Mask2 = POWERalpha(2*Byte1)*S0+S1

$$\text{POWERalpha}(2 * \text{Byte1} + \text{Byte2}) + \text{POWERalpha}(3 * \text{BYTE2})$$

$$\text{Mask1} = S0 + \text{Mask2} * \text{POWERalpha}(\text{Byte2}) / \text{POWERalpha}(\text{BYTE1})$$

In "C" these equations are written:

```

if (sum0 == 0)
{
  if (sum1 == 0)
    mask2 = 0;
  else
    mask2=power[255+alog[sum1]-alog[power[byte2+2*byte1]
      ^power[3*Byte2]]];
}
else
{
  if ((a=sum1^power[alog[sum0]+2*byte1]) == 0)
    mask2 = 0;
  else
    mask2 =
power[255+alog[a]-alog[power[byte2+2*byte1]^power[3*byte2]]];
}

if (mask2 = 0)
{
  if (sum0 == 0)
    mask1 = 0;
  else
    mask1 = power[255+alog[sum0]-byte1];
}
else
{
  if ((a=sum0^power[alog[mask2] + byte2]) == 0)
    mask1 = 0;
  else
    mask1 = power[255+alog[a]-byte1];
}

```

Notice that, in the code above, care is taken to check for zero values. The missing codeword position can be fixed by:

```

codeword[byte1] = mask1;
codeword[byte2] = mask2;

```

12.5. FEC Performance Considerations

The section above shows how to correct the different types of errors. It does not suggest how these corrections may be used in an algorithm to correct a bundle. There are many possible algorithms and the one chosen depends on many variables. These include:

- . The amount of processing power available
- . The number of packets per VBI to process
- . The type of hardware capturing the data
- . The delivery path of the VBI
- . How the code is implemented

As a minimum, it is recommended that the algorithm use single bit or single byte correction for one pass in each direction followed by packet replacement if appropriate. It is possible to do more than one pass of error correction in each direction. The theory is that errors not corrected in the first pass may be corrected in the second pass because error correction in the other direction has removed some errors.

In making choices, it is important to remember that the code has several possible states:

- 1) Shows codeword as correct and it is.
- 2) Shows codeword as correct and it is not (detection failure).
- 3) Shows codeword as incorrect but cannot correct (detection).
- 4) Shows codeword as incorrect and corrects it correctly (correction).
- 5) Shows codeword as incorrect but corrects wrong bits (false correction).

There is actually overlap among the different types of errors. For example, a pair of sums may indicate both a double bit error and a byte error. It is not possible to know at the code level which is correct and which is a false correction. In fact, neither might be correct if both are false corrections.

If you know something about the types of errors in the delivery channel, you can greatly improve efficiency. If you know that errors are randomly distributed (as in a weak terrestrial broadcast) then single and double bit correction are more powerful than single byte.

13. Appendix B: Architecture

The architecture that this document is addressing can be broken down into three areas: insertion, distribution network, and receiving client.

The insertion of IP data onto the television signal can occur at any part of the delivery system. A VBI encoder typically accepts a video signal and an asynchronous serial stream of bytes forming framed IP packets as inputs and subsequently packetizes the data onto a selected set of lines using NABTS and an FEC. This composite signal is then modulated with other channels before being broadcast onto the distribution network. Operators further down the distribution chain could then add their own data, to other unused lines, as well. The distribution networks include coax plant, off-air, and analog satellite systems and are primarily unidirectional broadcast networks. They must provide a signal to noise ratio, which is sufficient for FEC to recover any lost data for the broadcast of data to be effective.

The receiving client must be capable of tuning, NABTS waveform sampling as appropriate, filtering on NABTS addresses as appropriate, forward error correction, unframing, verification of the CRC and decompressing the UDP/IP header if they are compressed. All of the above functions can be carried out in PC software and inexpensive off-the-shelf hardware.

14. Appendix C: Scope of proposed protocols

The protocols described in this document are for transmitting IP packets. However, their scope may be extensible to other applications outside this area. Many of the protocols in this document could be implemented on any unidirectional network.

The unidirectional framing protocol provides encapsulation of IP datagrams on the serial stream, and the compression of the UDP/IP headers reduces the overhead on transmission, thus conserving bandwidth. These two protocols could be widely used on different unidirectional broadcast networks or modulation schemes to efficiently transport any type of packet data. In particular, new versions of Internet protocols can be supported to provide a standardized method of data transport.

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