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# AN OPTIMAL 'FULL-SERVICE' HFC NETWORK

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Abstract:

*Dual-cable (one cable upstream, one downstream) is recommended as the best solution to the upstream bandwidth problem. Separate power conductors provide flexibility and adequate capacity for powering the network and user interfaces.*

The fiber component is *`fiber as far as we can afford'.*

Optical power is expensive

- *'Linear'* optical power is much more expensive than *`digital'.*
- HFC networks will use `linear' (analog) optical transmitters. A transmitter/receiver pair costs about $15,000. This can be reduced to about $11,000 if fiber runs are shorter than maximum.
- A $15,000 optical TX/RX has a power budget of only 8 dB. This means that even if there were only short lengths of fiber between headed and viewer only 7 TV sets could be served -- about $2,000 per TV set! Under similar conditions a broadband RF amplifier has a power budget of about 40 dB. It could `light up' 10,000 TV sets.

RF power is cheap.

- Broadband RF amplifiers are in the $500 - $1,000 range and serve, on average, several dozen homes (taking into account cable losses in addition to power-division).

Fiber network component of HFC must therefore be amortized over a large number of customers.

- Co-ax network component of HFC is affordable down to level of individual customers

Fiber network is scalable - infinitely expandable in capacity:

Higher bit rates

- Wavelength-division multiplexing
- Additional fibers
- Spare fibers at installation
- Relatively easy replacement or reinforcement of optical fiber cables after initial installation.

The co-ax network is a serious bottleneck - not easily upgraded.

Additional (reinforcing) coaxial cables are expensive and difficult to install, mostly because of their bulk compared to fiber.

- Bandwidth is severely limited. Modem technologies face the *`Shannon information-theory'* limit. We are already packing almost as many `bits/Hz' as information theory allows. Co-ax doesn't have nearly as many useable *'Hz'* as fiber.

What is the practical bandwidth of a co-ax network?

Co-ax cable attenuation is approximately proportional to square root of frequency.

- This favors use of wider bandwidths.

Limitation is the repeater amplifier(s).

- Amplifier bandwidth is not a problem
- Amplifier distortion characteristic is the problem.
  - Distortion characteristics of amplifiers handling 'digital signals are different and more favorable than for all-analog signals.
- It is much easier to build a wider-bandwidth repeater amplifier for a network that is *'all-digital'* or *'mostly-digital'* than for an *'all-analog'* or *'mostly-analog'* network.
- Present *'catalog'* bandwidth is 750 MHz.
- 1000 MHz is readily available if a more realistic 'mostly digital' distortion spec' is applied.
- Even higher bandwidth (1500 MHz or even 2000 MHz) would be available within one or two years.
- Most repeater amplifier *'platforms'* are already spec'd to 1000 MHz.
- Many *'passives'* -- splitters, couplers, taps, etc. -- are already spec'd to 1000 MHz.
- 1000 MHz co-ax network will be 'easy' -- if the loading is *'mostly digital'*.

The coaxial cable bandwidth should be extended to 1,000 MHz. Paraphrasing the late Duchess of Windsor, ***"You can't be too rich, too thin or have too much bandwidth."*** Most *'legacy'* cable-TV systems are being rebuilt to 550 MHz bandwidth, a few to 750 MHz. 750 MHz is the highest *'catalog'* bandwidth from American manufacturers. New 'full-service' networks should be have bigger 'numbers' than the present 'heritage' systems -- more useful bandwidth and more electrical power.

Two-way (bi-directional) transmission in the co-ax network.

Alternative techniques:

- Frequency division
- Space-division (dual cables)

- Frequency division -- spectrum allocation -- *'upstream'*
  - 'High end' vs 'low end'.
    - *'Cable-TV'* systems are obliged to maintain FCC channels 2-13 (54-216 MHz) for analog TV (VSB-AM) channels (downstream).
    - This leaves practical options of:
      - 5-40 MHz (35 MHz) *(sub-low)*
      - 900-1000 MHz (100 MHz) *(high end)*

*'Low-end'* or *'high-end'* reverse path?

***'Low End'!!***

- 'High end' reverse path caps the forward path bandwidth.
- Once set it will be practically impossible to expand bandwidth in future.
- It is easier to predict reverse path bandwidth requirement than forward path.

    - If one of these paths has to be `capped' let it be the reverse path.
    - Filters in 900 MHz region waste a lot of spectrum (100 MHz) compared to filters at lower frequencies.
- 5-40 MHz isn't much bandwidth
    - Reducing node size helps, but is expensive - increases number of expensive optical TX/RX's
    - Improved access technologies (DAMA, etc.) increase utility of restricted bandwidth

The more reverse-path bandwidth per customer the larger our nodes can be:

- Major cost reduction - trades off expensive fiber network for cheaper coaxial cable.

**A better solution! -- Dual-cable!**

- A dedicated second co-ax cable for reverse path -- 0.625".

Co-ax cable attenuation (P-III) (dB/100')

| Size | 450 MHz | 550 MHz | 750 MHz | 1000 MHz |
|---|---|---|---|---|
| 0.625" | 1.30 | 1.45 | 1.72 | 2.03 |

- Trunk repeater spacing (22 dB) in 0.625" co-ax at 1,000 MHz would be 1,080'
    - Signal launch levels from subscribers' premises would not have to be excessively high because much of the attenuation in the reverse-path is due to *`passives'* which have *`flat'* loss characteristics.

Dual co-ax trunk-feeder-drop - to interface at customer `entrance'

- Various cable/wire options inside.

**Unrestrained reverse-path bandwidth allocation!**

**Unrestrained forward-path bandwidth allocation!**

Dual-co-ax networks are not uncommon.

- I built several large dual-coax networks in the early '80s.
    - Suburban Chicago - now owned by TCI and recently used for very successful NVOD/PPV trial.
    - Fairfax County (Virginia suburbs of Washington, DC) - now serves 240,000 subscribers.
    - 50 miles of dual-coax -- one forward and one reverse -- as adjunct to Fairfax County system

New `full-service' networks should be *`dual-cable'* -- two 1,000 MHz co-axial cables -- one for the `forward' path and one for *`reverse'* path. **The present 5-40 MHz *`sub-low'* reverse path allocation in a single cable is grossly inadequate. It is like building a super-highway with eight lanes in one direction and a dirt track in the other.** The ideal network has equal bandwidth in each direction. Who can reliably predict the degree of unsymmetry (if any) of traffic in these new networks? The best way is to provide symmetrical, adequate two-way capacity is with two cables - one for each service direction ..... 1000 MHz in each direction!

Alternatively, the second cable could be split to provide both forward and reverse bandwidth - `1-1/2' cables forward and `1/2' cable reverse.

Reverse path bandwidth examples:

*Sub-low* - 35 MHz - bandwidth per feeder:

| Com' Channels per feeder | BW per channel |
|---|---|
| 50 | 700 KHz |
| 100 | 350 KHz |
| 200 | 175KHz |
| 500 | 70 KHz |
| 1000 | 35 KHz |

*High-end* - 100 MHz - bandwidth per feeder:

| Com' Channels per feeder | BW per channel |
|---|---|
| 50 | 2000 KHz |
| 100 | 1000 KHz |
| 200 | 500 KHz |
| 500 | 200 KHz |
| 1000 | 100 KHz |

*Second cable* - 950 MHz - bandwidth per feeder:

| Com' Channels per feeder | BW per channel |
|---|---|
| 50 | 19 MHz |
| 100 | 9.5 MHz |
| 200 | 4.8 MHz |
| 500 | 1.9 MHz |
| 1000 | 1.0 MHz |

We could provide a dedicate 950 KHz channel full-time to each of 1,000 users! Multiple-access technologies, e.g. DAMA, allow even more users per node.

## MAXIMIZE DIGITAL TRANSMISSION

Digital transmission is better than analog in every respect. Digital picture quality is much better than with conventional `NTSC-analog' TV transmission. Minimize analog-TV services to minimum permitted by regulation. All `premium' TV services, i.e. those needing conditional access (addressable scrambling) should be digital. Hughes DirecTv has shown the practicability and viewer-appeal of digital TV transmission (my emphasis):

> *TELEVISION DIGEST via NewsNet*
> *Monday December 26, 1994*
>
> *DBS's 150-channel service "is a hoot," Washington Post TV critic Tom Shales said in ecstatic Dec. 21 report. Shales, who said he subscribes to both $29.95-per-month DirecTv and $34.95 USSB, called DBS "greatest new toy since the VCR, with* ***pictures so sharp and rich that it's as if a veil were lifted from in front of the screen. Cable cannot compare."*** *He praised on-screen program guide, but noted that service can be subject to rain fade and doesn't include local channels.*

## Electric Power In Co-Ax Networks

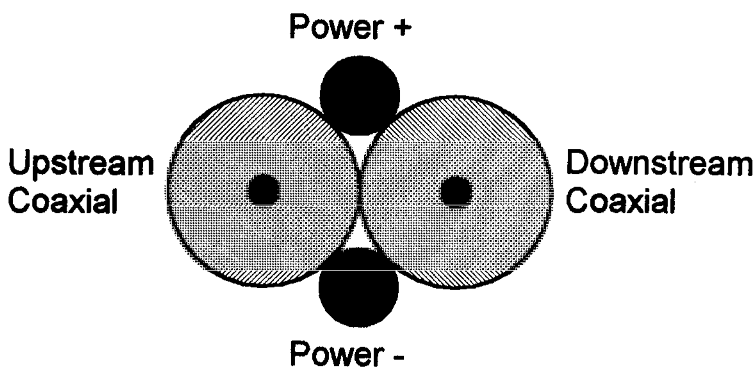
`Cable-powering' (60V AC on the center-conductor) is customary

- Problems:
  - Requires RF `chokes' in electronic equipment to separate power and RF. Difficult to provide good RF spec's when RF/power bypass is required.
- Power handling limitations because of relatively high resistance of co-ax cable center-conductor.
  - .625" cable - center conductor 0.136" diameter

- Al center -- 0.86 Ω / M'
- Cu center -- 0.55 Ω / M'
- High loop resistance limits ability to power supplemental network equipment, such as interfaces at customer premises entrance.
- Aluminum cable material creates electrogalvanic corrosion problems - necessitating use of AC.
- DC would be much more efficient.
- Low cost standby power provisioning

Separate (copper) power conductor would solve these problems.

- DC operation is feasible.
- Easy standby power provision.
- No electric power in RF carrying cables and components.
- Very low ohmic resistance. Conductor to at least #2 (AWG) size is practical (0.156 Ω / 1000').
- Moderately expensive - but worth it. A #2 copper wire (200 lbs of copper / 1000') costs almost as much as a .625" coax cable.
- Would allow network powering of user-interface equipment.




CABLE BUNDLE

# Audio Loudness and Dynamic Range in the Compressed Digital Environment

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Abstract

Digital audio compression systems are capable of delivering audio into the home or the headend with well over 100 dB of dynamic range. Will this potential dynamic range be exploited or squandered? This paper shows how features provided by the AC-3 audio compression standard allow a single encoded bit stream to supply an appropriate dynamic range to all listeners.

## INTRODUCTION

Current NTSC broadcast practice is often to provide audio suitable for the worst case listening environment (audio received via demodulation of RF and reproduced by tv with a built in loudspeaker). The audio is highly compressed, and the level of the highly compressed speech is such that peaks often hit 100% modulation. There is little headroom for left for music and sound effects to make a dramatic impact. The situation is somewhat better for feature films delivered by premium cable movie services, where less compression is applied, and some headroom above dialogue level does exist for music and effects. However, in order to leave some headroom the average level of dialogue must be reduced compared with that of broadcast channels and this leads to some consumer dissatisfaction when level variations are encountered while channel hopping.

Future delivery systems will incorporate digital audio compression (bit rate reduction) systems which can provide very wide dynamic range audio (>100 dB). These new systems must be interoperable with current broadcast systems where common speech often reaches 100% modulation, and must be able to interface with a conventional TV set by means of a Ch3 RF signal. This leads to a quandary where either the newly available >100 dB of dynamic range will be squandered so that speech can be heavily modulated to match current broadcast practice (to achieve reasonable level matching on Ch3 while avoiding the potential of overmodulation), or the new services will leave an appropriate headroom for music and effects above the level of speech in (which case they will not match the level of speech of off-air broadcast stations).

Consideration of these practical problems has led to the development of solutions. The AC-3 audio compression standard[1,2] provides a number of syntactical elements in the encoded bit stream which provide practical solutions to many of the problems of audio delivery. Programs may be encoded with differing amounts of headroom and still reproduce at a consistent level. A Ch3 remod output may be provided which is comparable (in dynamic range and loudness) to current broadcast practice. A baseband audio output may be provided with a dynamic range which is somewhat compressed (suitable for most listeners). The listener may be provided control to optionally reduce or eliminate the amount of compression which was applied by the program provider. In this case the listener (perhaps a home theatre enthusiast) can choose to listen to the original soundtrack free of any dynamic range signal processing. In short, a common bit stream can be used to supply audio service to wide range of listeners with differing needs, without severely compromising the audio delivered to any particular group of listeners.

This paper will go into some detail on these practical problems, and how the features of the AC-3 audio coding standard may be used to solve them. Some product design guidelines will be provided.

## AUDIO PRODUCTION

This section will review production practice followed in the creation of the highest quality soundtracks - those for major motion pictures.

Current film sound practice is to produce a multichannel soundtrack in the so called 5.1 channel format which is the subject of an ITU-R Recommendation [3]. This format provides five full bandwidth channels for left, center, right, left surround, and right surround. An additional limited bandwidth channel (the 0.1 channel) is available for high level low frequency effects (LFE). The LFE channel is limited to 120 Hz, and is adjusted to reproduce 10 dB louder than one of the full bandwidth channels. Reproduction of the LFE channel, in the context of the home environment, is considered optional.

During audio production, audio recorders are set up with a *reference level*. There is a match between this reference level and a sound pressure level (SPL) in the mixing studio of 85 dB SPL. That is, bandwidth limited pink noise, recorded at reference level, will measure an SPL of 85 dB on a sound level meter placed at the location where the mixing engineer sits. The sound recorders, depending on their type, typically have 20 dB of headroom available above reference level. Analog recorders using Dolby SR noise reduction have a soft overload characteristic and typically provide slightly more than 20 dB of headroom. Digital recorders, with their inherent hard overload characteristic, typically have exactly 20 dB of headroom.

The overload point of the audio recorders establishes a maximum SPL level for each individual channel. This level is (assuming 20 dB headroom above reference level) 105 dB SPL for each individual full bandwidth channel, and 115 dB SPL for the LFE channel. Assuming power addition in the room, all main channels driven to maximum level would result in an SPL in the room of 112 dB.

### Film Release Formats

The 70mm magnetic 6-track film format, using Dolby A noise reduction, can essentially deliver a film sound production to a theatre without compromise. (Actually there is some compromise at the frequency extremes where headroom on the 70mm print is reduced.)

The 35mm stereo-optical film release format has severe constraints and cannot deliver the full soundtrack to a theatre. This format only delivers two discrete channels of sound. A 4-2-4 matrix system is employed to produce the illusion of four channels (left, center, right, and surround). Headroom above reference level is only about 6-8 dB when Dolby A noise reduction is employed, or about 9-11 dB when Dolby SR is used. A substantial amount of peak limiting must be applied to make the original 5.1 channel sound production fit into the 35mm optical sound track. First, each individual track is capable of overloading the 35mm format. Second, when the tracks are mixed down (and matrix encoded) to the final matrix encoded two channel LtRt signal, there is gain in the potential peak level due to the fact that channels are being combined. In the theoretical worst case, about 20 dB of limiting can be required to produce the 35mm version of the LtRt from the discrete 5.1 channel master. In typical operating practice the worst case signals do not occur and only about 10 dB of peak limiting is actually required.

The AC-3 digital audio compression system was designed to encode the full dynamic range of a 5.1 channel film soundtrack into a data rate of 320 kb/s (although occasional peak limiting is necessary when encoding a soundtrack which has signal peaks in excess of +20 dB re reference level.) The Dolby SR• D digital film system was designed to place this data optically onto a 35 mm film print, and allow the full sound field to be decoded and reproduced in the theatre.

Home Video Formats

Video media differ in their sound capability, although today they are all limited to two channels. The consumer may choose to employ a matrix surround decoder in order to decode the delivered LtRt two channel signal into four channels.

When film soundtracks are provided for video media, the amount of limiting applied when the 2-channel matrix encoded LtRt signal is produced varies. In some cases the same version that was used for the 35mm film release is reused. In other cases, the transfer may be redone with a different amount of high level limiting, and perhaps some low level compression which brings up the loudness of the quietest parts of the soundtrack. The techniques employed depend on the intended purpose of the soundtrack. Is it intended for broadcast television? For a premium cable movie service? For an airline? For VHS (Hi-Fi tracks or linear tracks), or LaserDisc? Each media and customer may have a different requirement or request.

When the 2-channel LtRt signal is actually used for a home video format, further signal processing may occur. One example to explore is network broadcast television. The LtRt signal provided to a broadcast network may be (based on network request) highly compressed. When this signal is broadcast, the network will pass the signal through a signal processor in order to protect against overmodulation and to "improve" the signal for their audience. When the signal arrives at the local affiliate station the signal will again be passed through a signal processor to prevent overmodulation and to "improve" the signal for their audience. The result is often that the signal received by the poor consumer has been so modified that it bears only a faint resemblance to the original soundtrack which was so painstakingly produced.

THE AUDIENCE AND THE SET-TOP BOX

The problem in audio delivery to a diverse audience is that different members of the audience have different needs. This is in contrast to the original goal of a film sound production which is to produce a soundtrack destined for reproduction in the cinema to a captive audience. The original soundtrack needs to be altered to be useable by most consumers in a home environment. Unfortunately, conventional delivery systems (as well as some new ones) require that a single version of the audio program serve the entire audience, even though different members of the audience have differing wants and needs.

This section will look at the problem from the point of view of an audience member receiving service from a set-top box decoding a compressed digital service employing Dolby AC-3 audio coding. The audio component of this service may contain all of the 5.1 channels of sound originally produced. The set-top box will contain an AC-3 audio decoder which decodes the audio into an appropriate 2-channel stereo representation.

The audience will be segmented based on the type of interface used between the set-top box and the audio/video reproduction equipment. Three types of interface will be considered: RF remod on Ch3; audio line level baseband; and raw audio bit stream.

RF Remod on Ch3

Program channels will be received by the set-top box either as conventional NTSC signals (the off-air channels) or as digitally compressed signals. The off-air NTSC channels can be frequency converted to Ch3. If an off-air channel has stereo audio the RF output of the set-top box will contain stereo audio. The digitally compressed channels will be modulated on Ch3. In this case the RF output will invariably be monophonic due to the high cost to generate a BTSC stereo signal.

The consumer using the RF output will thus only receive monophonic sound on the digitally compressed premium channels. This monophonic sound will generally be reproduced by the loudspeaker inside the TV set. It is appropriate that this audio be quite compressed. The speakers inside a TV set are generally not capable of reproducing audio very loud, and so the volume will rarely be turned up very high. It is desirable that low level sounds not be reproduced so quietly that they are lost in the background noise typical of a home. The modulated level of dialogue in the RF signal should be close to that of off-air channels. This allows channel hopping without wild level fluctuations. The peak modulation level allowed may be 6-7 dB over 100% since a TV set can accept this level of audio overmodulation, and the audio modulation level on the output of a set-top box is not subject to regulation.

The type of audio appropriate for this consumer is similar to that appropriate for broadcast, except that peak levels may be permitted to be 6-7 dB higher. Loud sounds may (fortunately) be subject to less peak limiting than in the case of off-air broadcasts. Quiet sounds have a similar need to undergo low level compression (where the quieter sounds are brought up in level to improve their audibility). The level of speech should match that of off-air broadcasts.

Baseband Audio Output

Many consumers will wish to enjoy higher quality audio and video than can be supplied over the RF output. These consumers should be encouraged to access the baseband audio and video outputs. The baseband audio output will be a 2-channel stereo signal which, in the case of a 5.1 channel broadcast, may be matrix decoded into a 4 channel signal. This output will generally be connected to a much higher quality sound reproduction system than that found in a simple TV set. This sound reproduction system may include a matrix surround sound decoder, and may be capable of playing as loud as a cinema. When receiving an off-air channel, the baseband output will invariably be monophonic due to the cost to decode a BTSC stereo signal (although some manufacturers might supply BTSC decoding capability as an option).

The audio delivered out of the baseband audio output is not subject to the modulation limits which apply to the RF output. It is still desirable for the level of speech to match between off-air broadcast channels and digitally compressed channels. The lack of a modulation limit means that audio on digitally compressed channels can potentially be reproduced without any peak limiting. The fact that the audio reproduction system connected to the baseband audio output may be able to play very loud means that it is useful, for some of the listeners some of the time, to be able to deliver the soundtrack free from low level compression. However, most of the listeners most of the time will not be reproducing the soundtrack at anywhere near its intended (by the original mixers mixing for the cinema) loudness. Thus some peak limiting and low level compression is still very useful on the baseband outputs. The limiting and compression do not need to be as severe as in the case of the RF output and, ideally, they should be selectively defeatable by the home theatre enthusiast who wishes to enjoy the sound in all its intended glory.

### AC-3 Bit Stream Output

Some channels will supply digitally encoded audio bit streams containing the full 5.1 channel audio mix originally produced for the cinema. It is not anticipated that any set-top box decoder would ever provide all of these channels as discrete outputs as that isn't necessary. The consumer who wishes to enjoy multichannel surround sound will have a surround sound decoder. Today these units invariably contain a Dolby Pro Logic decoder which can decode the LtRt matrix encoded stereo signal. The set-top box can supply such a matrix encoded signal when receiving a multichannel audio transmission.

While the set-top box can supply a 2-ch signal on the analog baseband audio outputs, it can also supply a 2-ch PCM digital signal on an S/PDIF (IEC 958) digital audio output. This type of output (commonly found on CD players and DAT recorders) is preferred in order to deliver a 2-ch signal into a matrix surround decoder which employs digital signal processing, as it avoids an unnecessary stage of D-A and A-D conversion.

Today the home entertainment audio visual (A/V) receiver provides surround sound by means of matrix decoding of 2-ch signals. In the near future (late '95) models of these receivers will become available which also can decode raw AC-3 encoded bit streams into discrete 5.1 channel signals. The AC-3 bit stream interface on these new A/V receivers will be a modified form of the S/PDIF interface. This version of the interface is capable of conveying either the raw AC-3 bit stream or 2 channels of PCM audio. A particular input on the A/V receiver will thus accept analog 2-ch audio, digital 2-ch audio, or a raw AC-3 bit stream.

It is appropriate that the set-top box be capable of delivering either the decoded 2-ch PCM signal or the raw AC-3 bit stream out of the single S/PDIF output pin. The consumer using this output would (potentially) have access to all features provided by the AC-3 bit stream. The consumer using the AC-3 bit stream output will want access to the original soundtrack exactly as it was produced. However, like the consumer using the baseband output, much of the time this user will wish to reproduce the audio at a lower than intended level, and will thus wish to take advantage of peak level limiting and low level compression.

## THE AC-3 SOLUTION

When the AC-3 coding system initially developed for the cinema application was adapted to a form suitable for consumer use, the practical and differing needs of consumers were carefully evaluated. Early on in the development of the consumer version of AC-3 the designers decided to create a signal representation syntax which would allow a single encoded bit stream to be decoded into a form useable by nearly every potential listener. Factors which were considered included:

- How each listener might receive the audio signal.
- How many loudspeakers each listener might have.
- The dynamic range available on an RF interface link.
- The dynamic range which is subjectively desired.
- Level matching between programs and channels.
- Absolute traceability back to original mixing room setup.

### Decoder Downmix

When a 5.1 channel audio program is reproduced over a monophonic or stereophonic reproduction system, it is necessary to produce a 1 or 2 channel downmixed version of the original signal. The downmix can occur at the encoder if one is willing to employ matrixing techniques. This is the approach which was taken by MPEG multichannel audio, the so-called backward compatible (BC) system.

In the BC approach adopted by MPEG a multichannel audio transmission consists of two components. The first component is a 2-ch downmix. The second component is a set of three additional signals. The stereo decoder only has to deal with the first component. The multichannel decoder has to decode both components and, by means of a linear matrix operation, combine the two components to realize the original 5 channels. The approach is actually a 5-5-5 matrix system, where 2 of the intermediate 5 channels is the 2-ch downmix.

If the BC approach is used, all receivers end up with the same identical downmix. This can be a problem as different listeners will require different downmixes. Listeners who will be Pro Logic decoding a 2-ch downmix into 4 channels need a matrix surround encoded downmix. Listeners who listen in conventional 2-ch stereo or mono will benefit from a different type of downmix. In particular, if the mono listener receives a downmix intended for use by a matrix surround decoder, the information from the surround channels will not be present at all! Another problem with the BC approach is that the matrixing technique employed reduces the coding gain so that (as shown by the MPEG multichannel audio tests) a higher bit rate is required to achieve a given level of audio quality.

AC-3 is a non-backward compatible coder (NBC) which simply encodes all 5 channels. This allows the appropriate downmix to be performed in the decoder. Several types of downmix are available. One downmix is suitable for Dolby Pro Logic matrix surround decoding. Another available downmix is suitable for conventional stereo or (upon further summation) mono reproduction. This downmix features adjustable downmix coefficients so that the precise downmix can be adjusted to best suit the particular program which is being broadcast. The values of these coefficients are selected by the program provider and are delivered in the AC-3 bit stream to the decoder. The intent is to allow the program to be mixed for optimal 5.1 channel reproduction, and then to adjust the downmix coefficients for optimal stereo or mono reproduction. This is different from practice today, where a matrix surround mix is often severely compromised in order to adjust the result obtained in a mono or stereo reproduction of the mix. Today there is no way around the need to compromise; tomorrow, with AC-3, there will be.

Another problem with an encoder downmix has to do with the peak level buildup which occurs when signals are combined. Assuming one is trying to achieve some sort of level uniformity (where the speech is encoded at a common level) there will be a fixed amount of headroom available. When multiple channels of sound are downmixed into two channels, the potential peak level increases. If the available headroom is fixed, peak limiting will be required. This limiting will alter the signal and the original program dynamics will be lost.

In the case of the MPEG BC coder, if peak limiting is applied to the 2-ch compatible signal, in order for the full 5-ch decoder to unmatrix the 5-5-5 matrixed signals back into a discrete form, the same peak limiting must be applied to the 3 additional channels. This ruins the potential for the stereo or multichannel listener to ever exactly recover the original sound track dynamics. If a large amount of headroom is provided so that the peak levels after downmixing may be handled by the coder, there is still the problem of the RF remod with its limited headroom and the desire to match the modulated level of off-air speech. Either the level of speech will be much lower compared to off-air, or the peaks will severely overmodulate the remod, or a peak limited soundtrack must be supplied to the entire audience. The MPEG BC coder is constrained such that it cannot supply a proper version of the soundtrack to a wide audience.

While the AC-3 system does have a potential problem with the peak level of the 2-ch downmix occurring in the decoder vs maintaining a match to the off-air speech level, an elegant solution is provided in conjunction with the level normalization and compression control signals (compression now meaning level compression not bit rate compression).

## Level Uniformity

We define level uniformity in terms of human speech spoken in a normal tone of voice (neither shouting nor whispering). When one hops between channels, or one program segment ends and another begins, there is typically someone talking. The subjective level of that speech should be a constant, or else we will be annoyed by the level fluctuations.

One way to establish level uniformity is to require every program to encode speech at a common level. This requires broad agreement and strict enforcement (i.e. this is impossible). Another method is to allow speech to be encoded at any level (which is easy) but to indicate in the bit stream the level at which speech had been encoded (which is possible). This allows different programmers to operate differently and simply asks them to tell us, by means of a value in the bit stream, how they operate. The AC-3 bit stream contains a bit field which allows the level of encoded speech to be indicated. This value is used by the decoder to adjust the reproduction level. The result is that all programs and channels may be reproduced with a common level of dialogue. In the case of the Ch3 remod, this allows the modulation level to be set so that the modulation level of speech matches that typical of off-air channels.

## Dynamic Range Compression

The AC-3 bit stream contains an encoded representation of each individual audio channel free from any signal processing such as limiting or compression. No multichannel matrix processing is employed. The dynamic range of the encoded signals is much larger than desired by much of the audience much of the time.

The bit stream also contains compression control signals. These signals may be used by the decoder to reproduce the audio with an altered dynamic range. The control signals are generated by an intelligence up stream of the decoder. One type of control signal is generated to control the level of a mono downmix intended to be used by a 75 μsec emphasised FM modulator. The other type of control signal is intended to be used to provide an artistically reduced dynamic range signal for a 2-ch or multichannel reproduction using the baseband or bit stream interconnect.

## Compression for Ch3 Remod

The AC-3 bit stream contains a control signal (called *compr*) which is used when the decoded audio is intended to be modulated onto an FM carrier. The compr control signal is generated to assure that peak modulation is controlled to an acceptable value when speech is modulated to a value consistent with off-air broadcast channels. The use of this control signal by the decoder in the set-top box is well defined.

The compr control signal generator algorithm takes into account the indicated dialogue level, and assumes 75 μsec emphasis in the modulator along with a decoder downmix to mono. The algorithm may be told the acceptable peak deviation (typically 6-7 dB over 100%). The decoder, when optimizing the RF remod output, will use compr and the static dialogue level indication to adjust the decoder output level into the FM modulator. The result will be that the modulation level of speech will match that of the off-air channels, and peak deviation will not exceed the acceptable level. This will be the case no matter how many channels of audio are included in the bit stream. Of necessity, the amount of peak limiting (indicated by compr) which will be applied in the decoder will increase as the number of audio channels is increased.

The consumer using the Ch3 RF output of a set-top box is the type of consumer who will appreciate receiving audio which has had relatively more low level compression applied. Besides indicating gain reduction to prevent high level signals from causing overmodulation, compr can indicate gain increases which may be used to bring very low level sounds up in level. Thus the signal delivered at the Ch3 RF output may be tailored for the portion of the audience using that output, without affecting the signal delivered to other members of the audience.

Artistic Compression

Members of the audience who use either the baseband output or the bit stream output will typically not wish to be subjected to the full dynamic range of the original soundtrack, but will not wish to have the dynamic range restricted as tightly as is done by compr. AC-3 has a second dynamic range control signal (called dynrng) which is intended to be used by this portion of the audience.

The dynrng control signal indicates to the decoder that the signal level should be modified by up to ± 24 dB. This control signal is generated to provide a pleasing amount of gain reduction for loud sounds (those above dialogue level). Thus the maximum reproduced loudness may be controlled. For quiet sounds (those below dialogue level) dynrng can indicate a gain increase. This will prevent quiet sounds from becoming so quiet that they will be lost in the typical ambient noise level of a domestic listening environment. When not generating an output suitable for the Ch3 RF output, the AC-3 decoder will, by default, use the dynrng control signal and reproduce a restricted dynamic range as intended by the program provider. The implementation of the decoder may provide the individual listener the option to disable, in whole or in part, the use of this control signal. If the control signal is disabled in the decoder, the full dynamic range of the original soundtrack will be reproduced. The dynrng control signal can also be attenuated, so that is only has a partial action. If it is set for 50% scaling, then an indicated gain reduction of 10 dB will result in an actual gain reduction of 5 dB. The scaling can be adjusted differently for values of dynrng which indicate gain increases and those which indicate gain reduction. For example, values which indicate gain reductions could be scaled by 80% while those which indicated gain increases could be scaled by 30%. This would result in loud sounds reproducing with slightly less gain reduction than indicated (a 10 dB indicated reduction would become an 8 dB gain reduction), and quiet sounds being reproduced with much less gain increase than indicated (a 10 dB indicated gain increase would become a 3 dB gain increase). The program provider provides a control signal suitable for the mass audience. Individual audience members may select to adjust the reproduced dynamic range to suit themselves. The home theatre enthusiast is able to reproduce the original soundtrack exactly as it was produced.

There is one limitation to the user control of the use of dynrng. When the decoder is receiving a multichannel audio program and performing a downmix a to 2-ch stereo signal, there is the potential of overload due to channels being combined. For multichannel audio programs, the value of dynrng will be generated such that adequate gain reduction will be indicated to prevent overload in the downmix. In this case, the optional scaling of values of dynrng indicating gain reduction is prohibited in order to assure that the 2-ch downmix does not overload.

Audio Production Information

The most accurate reproduction of the original soundtrack occurs in the mixing studio where it is produced. After all, that is where the artistic decisions are made which result in the final soundtrack. For the soundtrack to be accurately reproduced, the sound reproduction system must match that of the original mixing studio, and the soundtrack must be reproduced at the same loudness (volume setting) used in the mixing studio. To facilitate this, the AC-3 bit stream may carry information about the mixing room characteristics and the volume level of the original mix.

The sound reproduction system of a mixing studio will be calibrated one of two ways. Small studios mixing for reproduction in a small room are set up with a flat monitor characteristic. Large mixing rooms which produce soundtracks for the cinema are set up with a monitor characteristic known as the 'X' curve[4], which is has an approximate 3 dB per octave high frequency roll-off beginning at 2 kHz. Large rooms are equalized to the 'X' curve to make them subjectively match reproduction in small rooms which are equalized to have flat response. If the 'X' curve were ideal, then a movie soundtrack mixed in a large room could then be played back in a small room and the frequency response would appear to be correct. Unfortunately, the 'X' curve is not perfect, and when programs mixed to the 'X' curve in a large room are reproduced in a small room equalized flat, the sound is perceived as being slightly too bright. Thus some equalization may be required when soundtracks mixed in a large room are reproduced in a small room, or vice versa. The AC-3 bit stream can carry a bit field which indicates what room size was used to create the original mix. Any decoder has the option to use this information and implement appropriate equalization.

Sound is perceived differently as its reproduction volume is changed. Accurate (to the original artistic intent) reproduction is only possible if the final sound is reproduced with the same absolute loudness as that used in the original mixing session. The AC-3 bit stream can carry a value indicating the absolute loudness calibration of the original mixing room. By using this value, a sound reproduction system can automatically set the reproduction volume to match that of the original mixing session. If the listener chooses to reproduce the soundtrack at a lower level, appropriate equalization can be provided to compensate for the ear's amplitude dependent frequency characteristic.

CONCLUSION

There are a myriad of issues which must be confronted in order to deliver an audio signal simultaneously optimized for many different types of listeners. The AC-3 coder has taken the approach which provides the original soundtrack in a form which may be reproduced exactly as intended by the original mixing engineers. Extra control elements in the bit stream allow independent optimization of the sound reproduced by the TV set receiving an RF input on Ch3, or the hi fi stereo system receiving a stereo baseband signal. The full featured AC-3 decoder receiving the bit stream output can take full advantage of all features provided by the AC-3 bit stream. The dialogue normalization feature will end, once and for all, consumer complaints about uneven loudness between channels. The final result will be more enjoyment of the audio soundtracks by all listeners.

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[4] ANSI/SMPTE 202M 1991.

# AUTHORING SYSTEM FOR FLEXIBLE AND RAPID DEVELOPMENT OF ON SCREEN DISPLAY (OSD) APPLICATIONS

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*Abstract*

*The On Screen Display (OSD) used in Zenith's Cable TV set-top boxes utilize an Authoring System to allow development of control screens, data structures and state definitions. These, when combined with information provided from the RDBMS, define its display and behavior capabilities.*

*This paper provides an overview of Zenith's ScreenPlay(TM) Authoring System used for the development of downloadable dialogs. Also discussed are the components of the Authoring System with respect to the modular, object-oriented design, and the advantages and limitations of the implementation.*

## INTRODUCTION

Within the context of the cable set-top box environment, an Authoring System can be defined as a software system that helps developers create multimedia programs or presentations without requiring the painstaking skills involved in traditional programming[1]. Today's set-top boxes, in addition to enabling scrambling and authorizing of video signals, play a key role in providing for interactive TV and information services. An interactive OSD (On Screen Display) application, once downloaded to the set-top box, controls the display and behavior of the box, depending on keystrokes from the viewer and resulting events. Some of the key features provided by the application, (also referred to as a *Dialog* in the subsequent sections), are such capabilities as scanning the program listings, making one-button selections from the listings for display or later recording, etc. The OSD application may be designed with many different *look and feel* user interfaces, as required by the cable TV MSOs. An Authoring System that is user-friendly and flexible is needed to rapidly develop and maintain the dialogs.

The remainder of this paper provides, in order, an overview of the design and implementation of the Screenplay Authoring System. Key advantages and limitations of the project as implemented are described. This is followed by sample screens. Conclusions are then drawn with respect to the experiences of the developers in this effort.

## DESIGN AND IMPLEMENTATION

### Design Requirements

As the set-top boxes are provided with greater resources, such as a faster microprocessor, more memory, and more complex support circuitry, the capabilities of the software that drives them must grow with it. When the design process was started, it was decided that the following criteria were basic to a successful Authoring System (AS):

- *Ease of Use.* The dialog development environment must be easy to learn and use, requiring only rudimentary programming skills.
- *Standard GUI.* The AS must provide a Graphical User Interface (GUI) that complies with the Common User Access (CUA).

- *Modular Dialog Development.* Provision must exist to permit definition of both internal functions and access to external libraries.
- *Procedural Language.* The AS must provide support for procedural actions for data and screen manipulation. This is accomplished via a C-like script language.
- *Ease of Maintenance.* It must be easy to maintain the dialogs as well as the authoring tools.
- *External Dynamic Data Access.* Provision must exist to permit reference to external dynamic data when developing a dialog.
- *Field Maintainability.* It must be feasible to hand over the dialog maintenance to the operators at the headend.
- *Authoring System Modularity.* The Authoring System components must exhibit a high degree of modular design to ease future enhancement.

## Implementation

The Authoring System is independent of any other component of the set-top box control system. As shown in figure 1, it has three active components, the Dialog Editor (**DE**), the Dialog Compiler (**DC**), and the Dialog Assembler (**DA**). These three components may be independently executed on different target platforms, if necessary[2]. Commonly, the DE is the only visible component; the DC and DA are transparent to the users. Dialog developers are only concerned with producing the dialogs, not with how the dialog gets translated to a form that the set-top boxes will understand.

The Authoring System produces, as its end product, a data store which describes the *static component* of the dialog; that is, the display and behavior of all data transmitted to the controller portion of the system. This data store is in a format suitable for input to the download process, which is part of the Information Gateway System. The download preparation process then combines it with the dynamic data from the SQL database(s) to produce a fully functional dialog with static and dynamic components, that is ready to be transmitted.

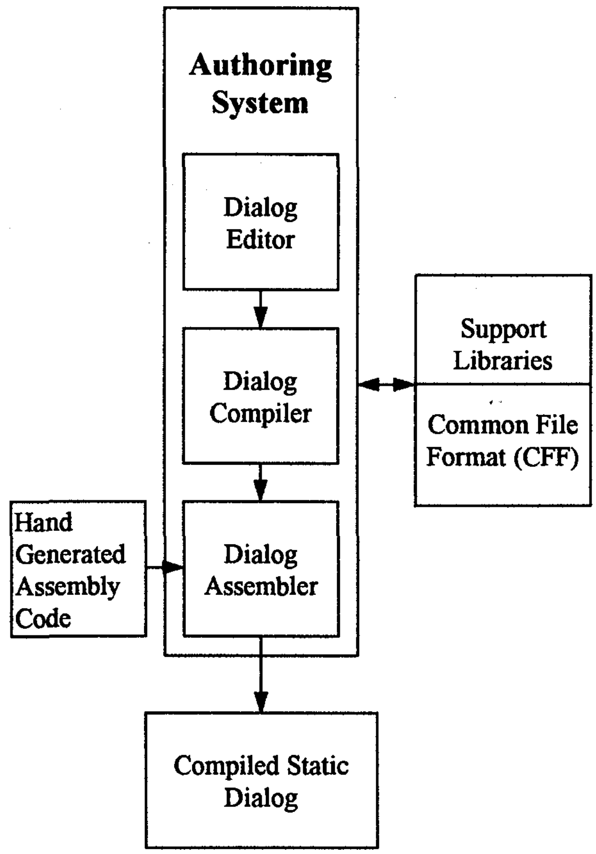


Figure 1. Authoring System

## Dialog Editor

The Dialog Editor is a WYSIWYG, ('What-you-see-is-what-you-get'), screen-oriented editor that enables the user to describe an interactive session in terms of multi-screen forms, display and input fields, and actions to be taken on user input or in response to events in the decoder or resulting from downloaded dynamic data and/or commands from the headend. The actions are expressed in a script language with a syntax reminiscent of 'C'. The output of this an editing session is a Dialog file.

At the time of the initial design, MS-Windows had established a widely installed base capable of running on minimal, standardized IBM personal computers; this drove the decision to implement the Dialog Editor as an MS-Windows application. Originally the entire Authoring System was implemented as a standard 16 bit application. Due to growing complexity and capabilities of authored dialogs, it has become necessary to convert the environment to a full 32-bit model. The eventual growth path is to convert to a full Presentation Manager (PM) application under OS/2.

Internally, each dialog consists of multiple sections, which in turn consist of multiple forms, in an hierarchical relationship. Each form is displayed to the dialog author as a graphic expression of what the viewer will see on the television screen, and set of transitions and/or actions that will be executed for each of the buttons on the decoder, or for desired events in the decoder, such as timers. The displayed form is created, designed by, and fully under the control of, the dialog author. The forms are treated as objects that consist of the display data (picture and literals), and the behavior, that is, the actions associated with each input key stroke or event. The user can define the action sequences for each of the input keys, as well as for predefined classes of events. For instance, this provides the flexibility of defining a single key to tune to a specific channel, start recording or any other actions that may be required; or to set an inactivity timer to return to a viewing state.

Due to the hierarchical nature of the dialog, sections, and forms, default behavior may be specified at each level, with local (i.e., form) responses differing from that defined at a higher level (i.e., section or dialog-wide).

Each dialog may have multiple action routines. An action routine is a common sequence of action statements. The action sequences can be in the form of the script language, which gets translated by the DC, or, it can have the native assembly source embedded for tasks too complicated to easily express in scripting, or to permit performance optimization. Despite the fact that use of embedded assembler code requires thorough knowledge and understanding of the assembler and decoder architecture, this capability provides the knowledgeable user with a great deal of flexibility.

A dialog can have internal action routines that are visible only to that dialog--either attached to a specific display object, or defined as callable procedures. External assembler action routines may be imported via prototype definitions. (Future versions of the DE will be capable of importing/exporting script routines.) This allows the creation of common libraries of useful routines usable by multiple dialogs, reducing the per-dialog unique development necessary for ongoing maintenance and authoring.

## Dialog Compiler

The Dialog Compiler accepts as input the binary data store created by the DE. It then reduces actions to executable code sequences, resolves the tokenized global and local data references from the intermediate form to absolute dataset type and field references, produces the final state transition dataset records, and emits an ASCII Dialog Assembler output data store. Any necessary information pertaining to external data references are provided by the Record Set Definition Export (RSDE) file.

The Dialog Compiler is wholly written in 'C' as a portable application--although currently an OS/2 application, it was successfully compiled and executed on UNIX platforms during its development. Its execution platform, and invocation, is usually totally transparent to the user. This is accomplished via a command interpreter script that compiles and assembles a dialog source file in a single invocation, much in the way the early 'cc' command in UNIX was implemented. This shell script invokes the appropriate components depending on the command switches, passing options along to the correct target component as necessary.

## Dialog Assembler

The Dialog Assembler accepts as input ASCII files expressed in terms of the Dialog Processor Assembler Source. This is in the form of a traditional assembler, e.g., opcodes, operands, labels, pseudo-ops, etc. Relocatable and link-editable formats are not supported; all references must be satisfied by local declarations or globals provided from the RSDE data.

It is implemented as a one-pass non-relocatable assembler; as such, all definitions and references must be contained in the primary and included files at assembly time. It supports an indefinite number of forward references, however, which are resolved via a patch look-up scheme as opposed to the more traditional two-pass approach.

Input to the DA is either the output of a DC execution--which is therefore the result of a DE session--or explicitly coded assembler source. If the latter, it is the responsibility of the author to provide for any external definitions required as described in the RSDE file. Explicitly coded assembler source requires thorough knowledge and understanding of the assembler and decoder architecture.

Output is in the form of a binary file suitable for processing by the Download planning and transmitting components of the Information Gateway.

## Support Libraries

The three components DE, DC and DA share the information concerning the dialog and data. As a result, there are well-defined in-memory and data store structures for each of these interface requirements. These interface definitions are expressed via common support routine libraries provided to simplify processing of shared data and to assure identical views of such data.

The Common File Format (CFF) library allows creation of arbitrary data sections within a binary file; the contents of each section is unknown by, and irrelevant to, the library manipulation routines. Thus, this mechanism provides an operating-system independent program-level means of archiving information that is logically related, but must be handled in discrete separate sets.

The routines provided by the CFF library, as might be expected, are oriented to providing easily-used performance of the following tasks:

- Creating the CFF file
- Retrieving and manipulating existing file sections and data
- Adding and deleting file sections
- Providing information about the current file status and contents

## Dynamic Data

The RSDE library provides the callable routines necessary to extract the data associated with various fields, and to construct the ASCII datastore which permits export of this information to the Dialog Editor. It also provides support routines to import data from this file into 'C' structures, and to populate the DE symbol table.

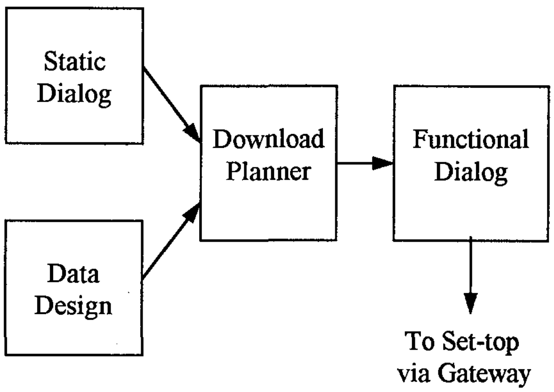


Figure 2. Preparation and Download

Data Design has the information on the dynamic data recordsets. Download planner, which is a part of the downloader, combines the static dialog created by the Authoring System, with the dynamic data and produces a downloadable fully functional dialog as shown in figure 2. The packet sequences are also prioritized.

## Authoring System in the Headend

The Authoring System is implemented as a sibling to the Information Gateway System at the cable plant headend, as shown in figure 3. Information Gateway System is the main link for the Authoring System. Detailed description of the Information Gateway System is beyond the scope of this paper.

Other data sources typically include program guide data and sports and weather information. Once a dialog is created by using the Authoring System, the compiled output is sent to the Gateway, which then gets the dialog prepared and starts transmitting to the set-top boxes.

As and when the Gateway receives any new data, which may be an entirely new dialog or new dynamic data, the downloader transmits the incremental update, which is totally transparent to the user. That is, the subscriber of the set-top box is not aware of when and how the data is received. The Gateway also provides the capability to altogether reset the set-top boxes and start afresh with a new dialog.

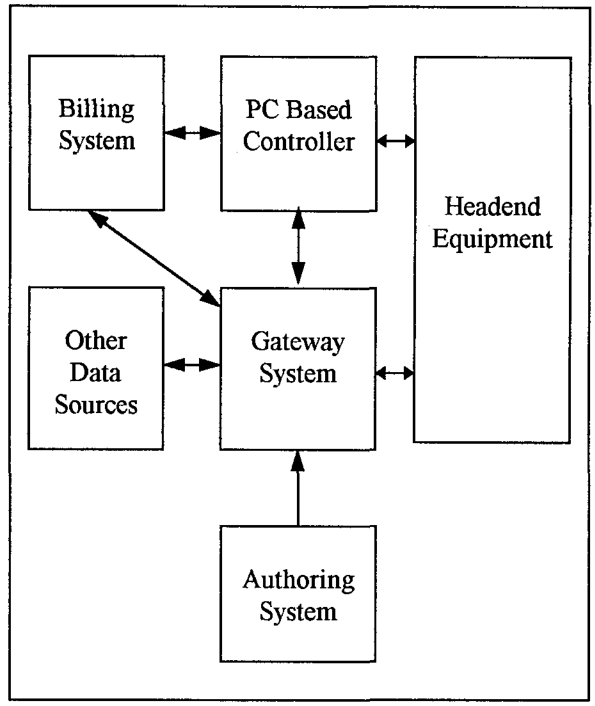


Figure 3. Typical Headend

## ADVANTAGES AND LIMITATIONS

As with any software system, the experience of development and use has reinforced some of the early design decisions, and pointed out some of the oversights and shortcomings in others.

### Advantages

- *Customer Acceptance*. We have had excellent response from our customers who are currently using the Authoring System.

- *Rapid Prototyping.* As hoped, even without underlying external data for dynamic displays, the tools have proven useful and usable by Sales or Marketing personnel. Non-programmers can, and do, create dialogs using their laptops running MS-Windows or OS/2 without having to compile and assemble the dialogs. These may be viewed for acceptance and modification by customers, and later compiled, assembled, and downloaded for on-set review.
- *Reduction in Required Developers Skill Sets.* Prior to the introduction of the Authoring System, dialog developers were required to be software developers. Experience has shown that personnel with minimal traditional software development experience can successfully produce functional dialogs. Despite some concerns that the state-driven model of the set-top decoder might present problems, this has not proven to be an obstacle. And for those tasks requiring greater skill sets, fewer skilled developers are required to intervene.
- *Flexibility.* The ready ability to define the keys to suit the needs of the headend, of different customers, and even of different dialogs for the same customer, has proved to be an easily exploited and useful capability. Rapid modification and extensibility of the dialog, even to the extent of major logic redefinition, has proved to be a benefit of the system.

## Limitations

- *Hardware Dependence.* Despite the intention of virtualizing the interrelationships between hardware and the Authoring System, there are still too many dependencies at the Dialog Editor level on underlying capabilities of the display hardware.
- *Need expertise in MS-Windows API to maintain the Editor.* The time needed to develop, debug, and modify the MS-Windows code proved to be one of the greatest bottlenecks during the development effort; plus, skilled MS-Windows programmers are at a premium. In retrospect, a solid GUI application generator with support for multiple platforms would have reduced the workload significantly.
- *No support for relocatable and link editable formats.* This was designed as a follow-up capability to the basic tool; 'hooks' were built into the initial release to facilitate this. Development of the first few dialogs pointed out the importance of this capability; it should have been an integral feature of the initial release.
- *Script Language Implementation Incomplete.* In a common problem with object oriented systems allowing attachment of script or code to objects, it's difficult for authors unfamiliar with a dialog to garner a complete picture of the actions embedded in the dialog without extraction and printing tools. We realized the lack of this feature and plan to implement it in the upcoming version.
- *Lack of Emulation.* The debugging environment received far too little attention in the original plan. Particularly painful in its absence is an emulation environment which would permit debugging of complex dialogs without assembling the panoply of equipment necessary to support a functional download of static and dynamic data. We plan to overcome this by adding an emulation feature as part of the Authoring System.

## SAMPLE SCREENS

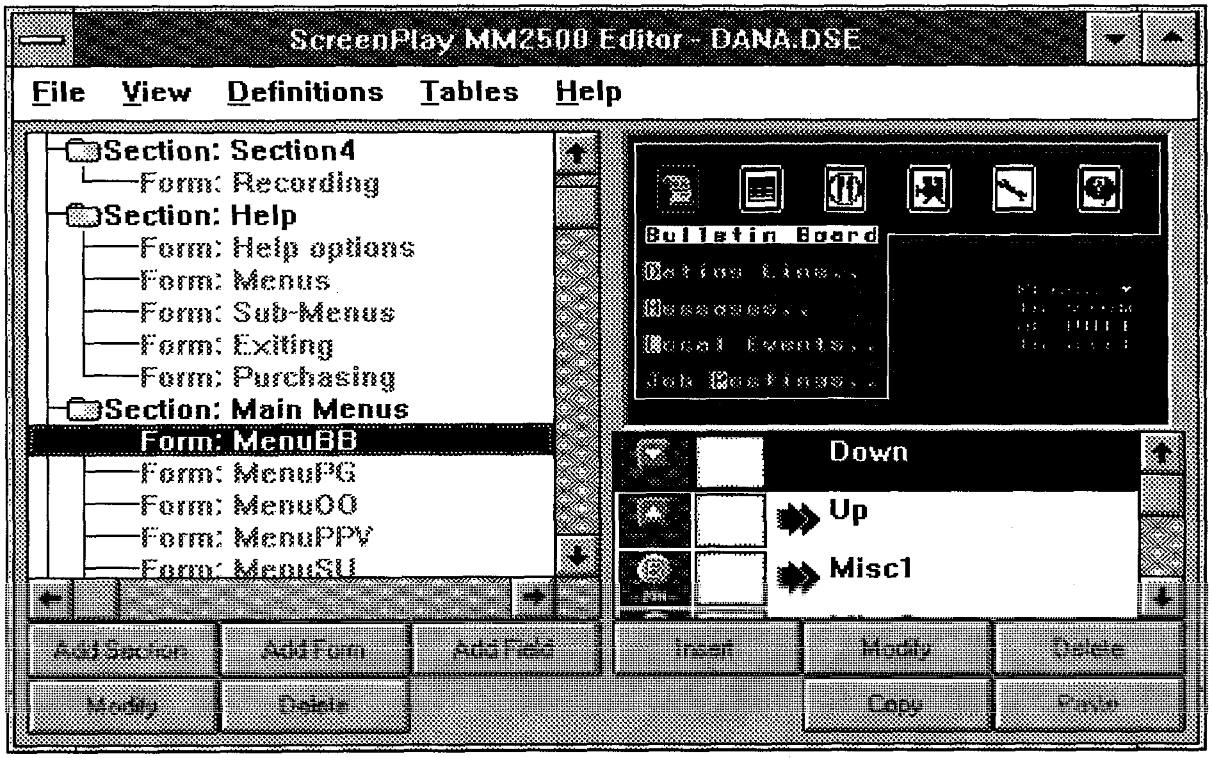


Figure 4. Main Screen

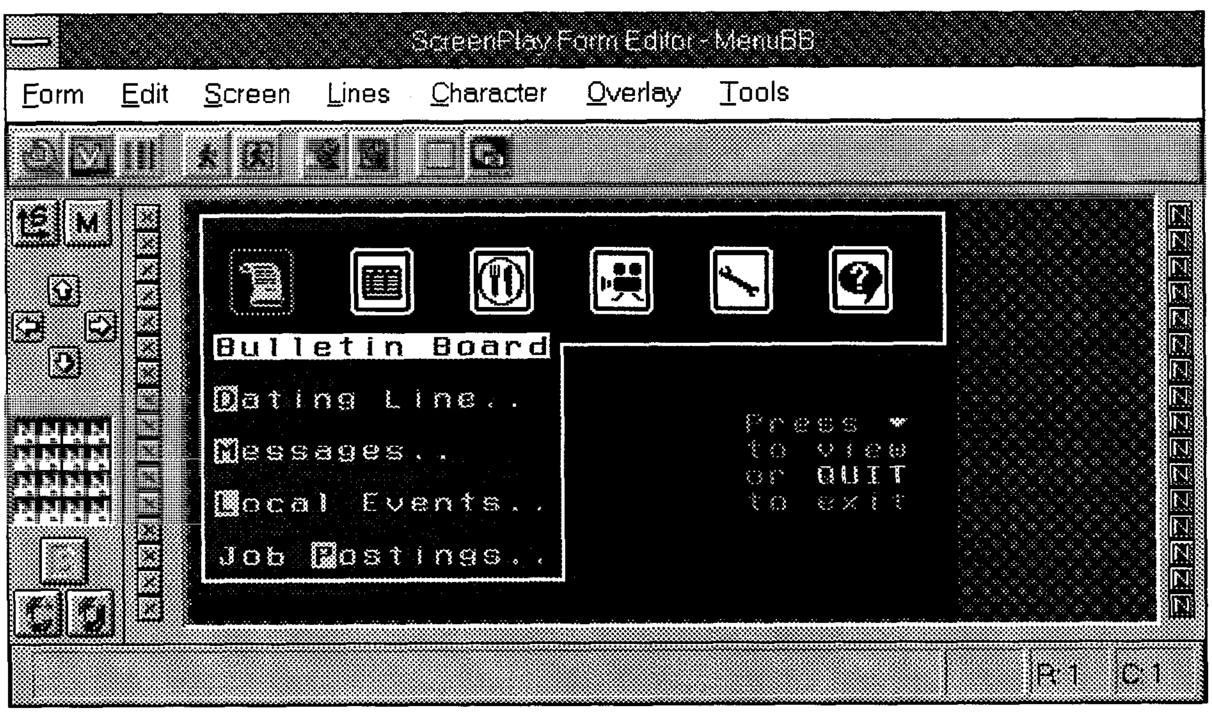


Figure 5. Form Editor

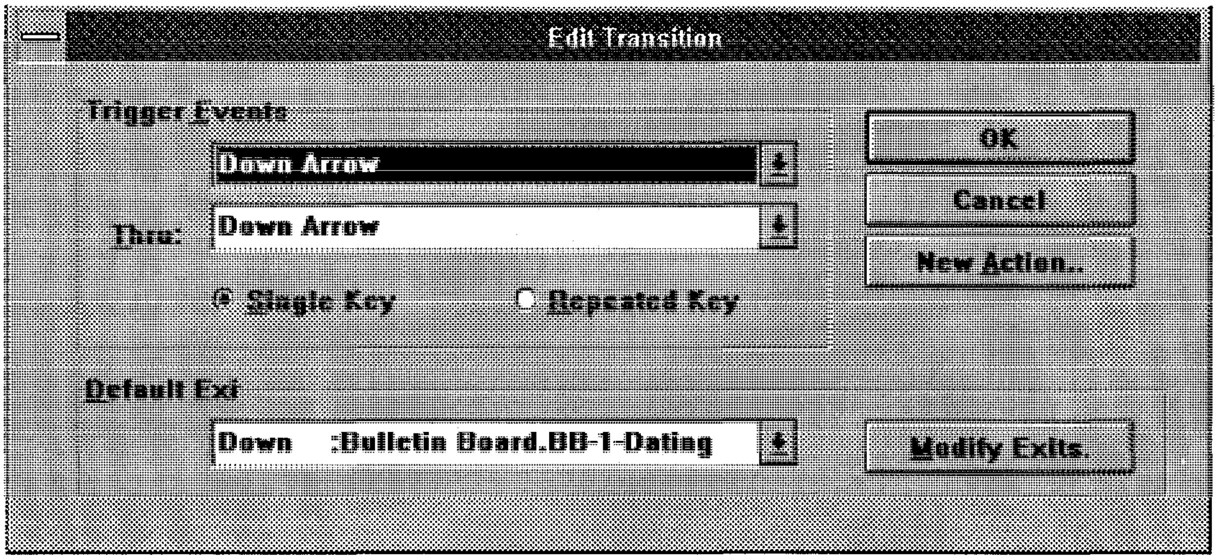


Figure 6. Actions for an Input Key

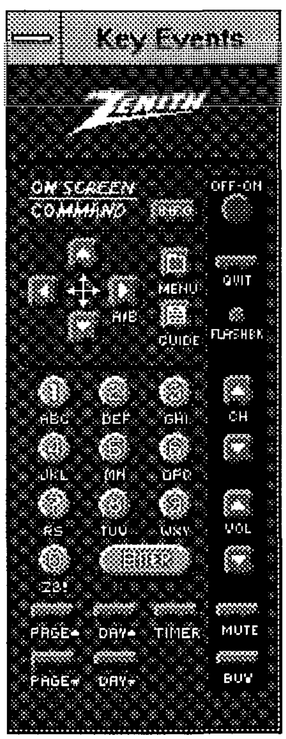


Figure 7. User Remote

Figure 4 shows the main screen of the Dialog Editor. It shows an opened dialog which has multiple sections and forms defined. The highlighted form MenuBB is being currently modified by the user. The user can design the screen display and define the transitions for the appropriate input keys. A transition is basically a sequence of actions to be executed for an input key. This may be as simple as switching to a different form or executing more instructions.

Figure 5 shows the form editor which lets the user paint the screen as it is intended to be displayed on the TV. The screen area is divided into multiple cells. Each cell may have its own attributes, like background color, foreground color, blinking etc. Each form may have the video on or off. Any lines on the form may selectively be enabled or disabled from being displayed.

Figure 6 shows the action(s) to be taken on a particular input key on the remote, in this case, the Down Arrow key on the remote. Figure 7 shows the replica of the remote control. The user just has to click on any key when defining an action sequence for that key.

## CONCLUSIONS

The design approach is sound. An Authoring System should enable the users with less programming skills and minimum training, to create dialogs. The Screenplay Authoring System which keeps evolving, does achieve this primary goal. In this implementation process, a significant amount of custom software for the tools was developed , a major portion of which can be reused. In retrospect, the use of some off-the-shelf software packages would have saved some time and effort.

It is obvious that the interactive TV is the wave of the future. It is going to be increasingly important to provide the tools that are easy and flexible to develop the interactive TV applications. It should be possible for marketing experts to decide what they want the TV screen to display without having to become technology experts. By handing over the maintenance of these dialogs to the cable TV MSOs, the system provides them with the control and flexibility.


## ACKNOWLEDGMENTS

Many thanks to our colleagues in the cable TV software engineering department for their review of the paper and valuable suggestions. Special thanks to Winston Tsao for his help in preparing the sample screens.


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# AVAILABILITY CALCULATIONS AND CONSIDERATIONS IN A BROADBAND HYBRID FIBER COAX NETWORK FOR TELEPHONE SERVICES

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## 1.0 ABSTRACT

*There is a growing acceptance for Hybrid Fiber Coax architectures to serve the many requirements of a multi-media environment. Given this application of analog video and telecommunications services, there is a great desire to understand and predict the network availability for telephone-like services. This paper will address the various elements involved in the actual offering of telephone service and their resulting impact on the availability of the network services. Efforts will be made to characterize the individual contributions as well as methods to improve the overall availability by use of techniques such as electronic redundancy and/or diversity. Projections will be made on availability of POTS services in that environment, given the configuration of the optics, the number of amplifiers in cascade, powering alternatives and other contributing factors in the availability of the service.*

## 2.0 AVAILABILITY CALCULATIONS

Clearly the most important question involving any system reliability involves the up-time of the system. This has significant implications over the operational cost of the service provider. The failure rate of the various components of a network has direct impact over the size of the spare stock. The level of training, the size of the maintenance staff and their geographic spread are all dependent on the diagnostic capability of a system and the frequency and location of necessary maintenance actions. Also, the number of service interruptions affect the level of satisfaction of a subscriber. It is therefore clear that a service provider would view the overall reliability and the up-time of the product as a key requirement.

The chance that the system will be up at any particular time, and the percentage of time the system is operating are two measures which reflect the up-time of the system. Availability of the system is defined as [2],

$A = \lim P(t),\ t \to \infty,$

where P(t)= "probability that system is operating at time t".

The unavailability of the system will be denoted by U, and is given by

$$U = 1 - A \qquad (1)$$

It can be shown that the long term availability performance of a replaceable system component is related to its long term Mean Time Between Failure (MTBF) and its Mean Time to Repair (MTTR) as

$$A = MTBF/(MTBF + MTTR) \qquad (2)$$

It can be observed that as MTBF increases, representing a more reliable subsystem or component, the availability A approaches 1.0, regardless of any finite MTTR. Also, as MTTR approaches zero, A approaches 1.0, regardless of MTBF.

Another common term in reliability analysis is, Failure In Time (FIT). FIT is the average number of failures in $10^9$ hours of operation. MTBF and FIT are inversely related as follows:

$$MTBF = 10^9 / FIT \qquad (3)$$

## 2.1 SYSTEM RELIABILITY MODELING

System reliability analysis is a time consuming endeavor requiring a wide variety of technical skills. The steps involved are [2]:

__1. System Definition__: The first step in performing a reliability analysis is to obtain a complete and precise definition of the system architecture from both a functionality and a reliability point of view. A graphical representation of the architecture such as reliability block diagrams or state transition diagrams is very helpful in providing a precise and usable description of the reliability architecture of the system. In the context herein, a system may be the Access network consisting of the end-to-end equipment from a class 5 switch to the home terminal equipment or could be a stand-alone rack of equipment, such as the Head-end Switch Interface Unit (HSIU); both are shown in figure 1.0.

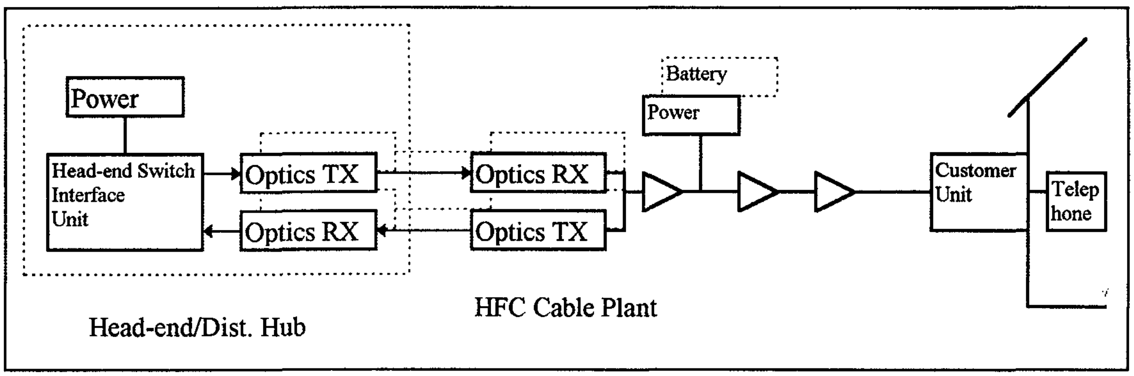



**Figure 1.0 Hybrid Fiber Coax Model for Telephony**

__2. Reliability, Availability, and Maintainability (RAM) measures selection__: Before a reliability analysis can begin, it is necessary to determine what measures of reliability, availability and maintainability are to be predicted. Care should be taken that the measures reflect the features of the system that are of interest to the service provider. The right selection of MTTR is very key in making the results plausible. The MTTR chosen for various parts of the system must reflect the average performance of a maintenance organization for the various parts of the system. Bellcore TR909 [3] suggests the following MTTR values:

| Location or Type of Equipment | MTTR (hours) |
|---|---|
| For Central Office or manned head-ends | 2 |
| For Host Digital Terminal (HSIU) | 4 |
| For Outside Plant equipment | 6 |

**Table 1.0 Mean Time To Repair Values for Fiber-In-The-Loop**

__3. Architecture Analysis and Decomposition__: To predict the appropriate reliability, availability, and maintainability measures for a complex system such as the HSIU, for which there are a large number of subsystem modular components, a tremendous number of states may be required to appropriately model the system.

__4. Architecture Modeling__: Architecture modeling involves creating reliability model(s) that describe the behavior of a system. Features such as interactions between the hardware and software components, maintenance and operational characteristics, and fault detection and isolation characteristics must all be taken into account. It turns out that the end-to-end system or network can be modeled as a chain of complex system components. Each complex network component in this chain can be analyzed rigorously and individually. The results can then be used to compute the end-to-end availability with much simpler methods. This can be done as long as the components are statistically independent.

__5. System component parameter determination__: One of the most important (and often time consuming) tasks in system reliability modeling and analysis is the determination of the parameters of the reliability model such as circuit pack failure rate, maintenance and repair times, and detection and coverage probabilities.

There are three methods to calculate availability parameters as described by Bellcore TR-332 [1]. These will be discussed in section 2.2

**6. Model Solution and computation of RAM measures**: Once the reliability model(s) are created and the parameters have been determined the model(s) can be solved for the appropriate RAM measures

**7. Model Parameter Sensitivity Analysis**: One of the usages of reliability analysis is the optimization of system architecture. Often this task takes the form of determining the effect of changes in a parameter on the desired availability measure, then subsequently finding the value that optimizes the measure. In these cases also, the effect of the choice of parameter value on the resulting availability measure is of major importance. A parameter sensitivity analysis is the procedure that assesses the effects of changes in the parameter on the desired availability (RAM) measures. *This procedure is especially appropriate for finding the weak link in an architecture. Also, through this process a development organization can determine whether they are over-engineering or under-engineering redundancy in a system.*

### 2.1.1 RELIABILITY BLOCK DIAGRAM

(COMBINATORIAL METHOD)

Reliability Block Diagrams (RBDs) are commonly used to represent the reliability architecture of a system. RBDs consist of a simple pictorial method to represent the effects of all possible configurations of functioning and failed components on the functioning of the system. RBDs are most useful when in analyzing the end-to-end availability of a transmission system such as the Hybrid Fiber Coax (HFC) architecture. This is due to the functional independence of the components of the end-to-end system. It will become clear that in a complex stand-alone system such as the HSIU, the State Space Method would be more appropriate. In the HFC architecture as shown in figure 1.0, there is no functional inter-dependence say, between the function of the Customer Interface Unit and the Fiber Optics, except for signal transmission processing. The Optics has to pass the broadband signal through, with some level of integrity. To demonstrate the point we assume that the optics either processes signal with no through degradation, or passes no signal at all and that is only upon unit failure. The CIU does not rely on the Optics to function properly. Because of the functional independence property, the failure analysis of the HFC architecture would most appropriately be analyzed using RBDs or using the academic terminology, the combinatorial method. To understand the availability of the HFC architecture, we must establish some simple combinatorial rules. It can be proven that the availability of a serial chain as shown in figure 2a, when the members are statistically independent, is the product of the availability of each chain member. Therefore the availability of the chain, when the availability of each component is known, and when they are statistically independent, is given by

$$A_{eq} = A_M * A_N \qquad (4)$$

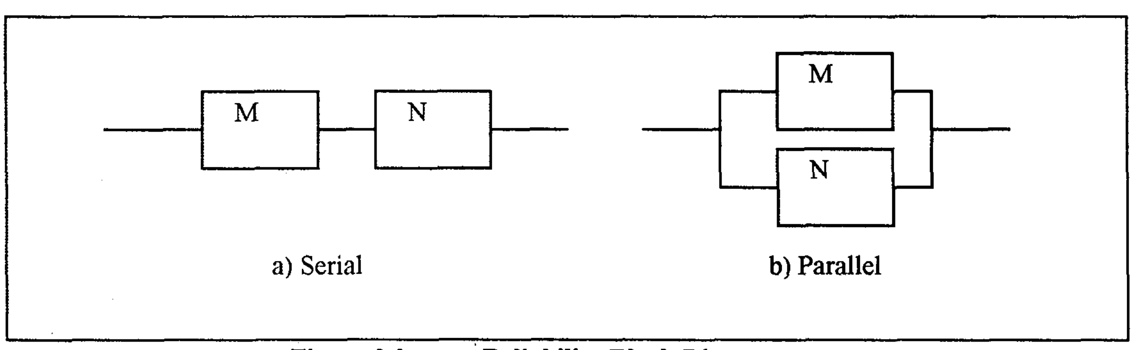



**Figure 2.0 Reliability Block Diagrams**

For the parallel blocks of figure 2b, the equivalent availability of the chain is given by,

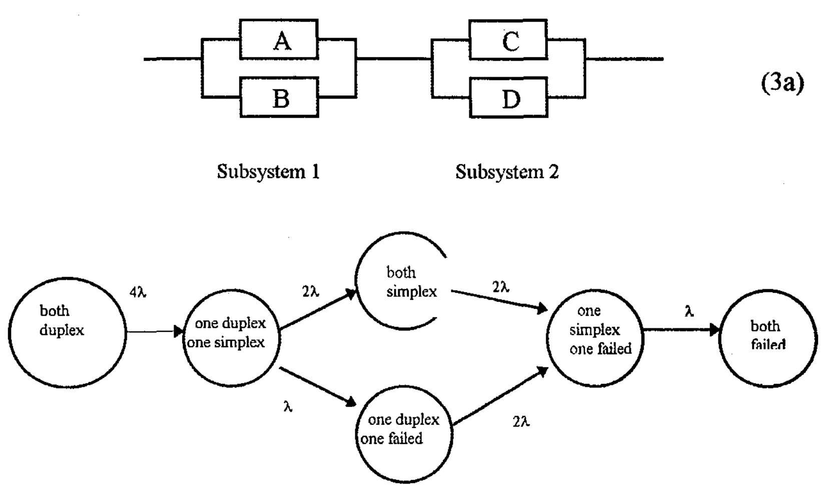
$$A_{eq} = A_M + A_N - A_M * A_N \qquad (5)$$

The relations of (2), (4) and (5) are sufficient to analyze any RBD availability. In section 3 we will apply these equations to the reference model of figure 1.0 to derive the availability of a typical HFC architecture, for the delivery of telephony services.

### 2.1.2 STATE TRANSITION DIAGRAM

(STATE SPACE METHOD: MARKOV MODELING)

For most complex systems, there are more modes of operation than simply just working or failed. There are a variety of different working modes where certain components have failed but are being covered due to redundancy, a variety of different failed modes where different components have failed and brought down the system, and other modes of degraded operation which cannot be conveniently labeled as working or failed. Each possible mode of operation for a system, and the set of states of the system is called the *"State Space"*. It is customary to assign the list of possible states of the system the numbers 1,2,....,n and refer to the model as an n-state model. A *"State Transition Diagram"* is a pictorial view of all of the possible operational states of the system, with arrows representing possible transfers (called state transitions) in the mode of operation of the system. In Figure 3.0 the two reliability modeling methods for the same system are illustrated.




(3a)

(3b)

The numbers shown on the lines are failure rates (commonly expressed in FITs). It is also assumed that the times between the occurrence of events are exponentially distributed. Exponential distributions have many useful properties. It turns out that an exponentially distributed process, call it X, of (failure) rate λ, is defined as

$$P( X>t) = e^{-\lambda t} \qquad \text{for } t >=0.$$

An exponential distribution, is fully defined once the rate λ is known.

In figure (3a) a system is shown using RBDs. The system consists of two identical subsystems in series with each subsystem having two identical components in parallel, labeled, A, B, C, and D. It is assumed that all components have exponential distributions with common failure rate λ. The same system is shown in figure (3b) using the state transition diagrams.

Some of the properties of exponential distributions have been used to go from the RBDs to the State Transition Diagram. For example $P(X_A \cup X_B > t) = e^{-2\lambda t}$ , where $P(X_A > t)$, represents the "time before failure" distribution for the component A. The $4\lambda$ rate shown on the line moving from "both duplex" to "one duplex, one simplex" state represents the equivalent rate for the union of the four processes A, B, C and D, since any singular failure in a both duplex state will result in a transition.

Once the repair statistics are incorporated into the model, then many Availability calculations can be done. For the sake of time, we will not get into this method any further, however, we will state the only proper way to understand a certain availability measure of a system or a subsystem with inter-dependencies is via the State Space Method or the Markov modeling method. Care must be taken that the behavior of the system has been properly modeled. It is very easy to overlook the fault detection and diagnostic portion of a system. Most fault tolerant complex systems utilize rather sophisticated fault detection, diagnostic and recovery schemes. The more complex these schemes are the higher will be the probability of fault recovery malfunction. This fault recovery scheme is handled through some hand-shaking between hardware and software. This complex interaction between software and hardware can only be modeled through Markov models. To apply combinatorial methods (RBDs) to these complex schemes will result in over-simplification and illusive conclusions.

## *2.2 SYSTEM ELECTRONIC COMPONENT RELIABILITY PREDICTION*

The majority of the failures in the field are due to a hardware component failure. Therefore, there is a need to predict beforehand or assess the field reliability performance of replaceable hardware assemblies.

The three methods in Bellcore TR-332 for calculating the MTBF of a system's replaceable components (plug-in, power converter module, etc.) are given in the following subsection. Method I, is basically a parts count method. Method II combines results of lab testing with Method I. Method III allows for incorporating field return data into the long term predictions. In this paper we will only look at Method I in some detail.

### 2.2.1 Method I

This is basically the parts count method. This method provides a starting point when no lab test results or field failure information are available. Bellcore defines the steady-state failure prediction of a replaceable system component, circuit pack, or assembly, as $\lambda_{SS}$ , where,

$\lambda_{SS} = \pi_E \sum N_i \lambda_{SSi}$ , with the summation done over i=1,..,n, and (6)

where

n = the number of devices in the assembly

$N_i$ = quantity of the $i^{th}$ device type

$\pi_E$ = unit environmental factor as explained in Table 2

$\lambda_{SSi} = \lambda_{Gi}\, \pi_{Qi}\, \pi_{Si}\, \pi_{Ti}$ , steady-state failure prediction rate for the $i^{th}$ device (7)

$\lambda_{Gi}$ = Generic failure rate for the $i^{th}$ device, given in Table 3

$\pi_{Qi}$ = Quality factor for the $i^{th}$ device, given in Table 4

$\pi_{Si}$ = Stress factor for the $i^{th}$ device, given in Table 5

$\pi_{Ti}$ = Temperature factor for the $i^{th}$ device, given in Table 7

| Environment | $\pi_E$ | Nominal Environmental Conditions |
|---|---|---|
| Ground, Benign | 1.0 | Nearly zero environmental stress, e.g. Central Office, CEV |
| Ground, Fixed | 1.5 | Conditions less than ideal, some environmental stress, e.g. manholes, remote terminal, customer premises subject to shock, vibration, or temperature variation |
| Ground, Mobile | 5.0 | Conditions more severe than previous row. Mobile telephones, test equipment |

**Table 2, Environmental Factor**

| Device Type | Failure Rate (in 10E9 hours) | Temp Stress Curve | Electric Stress Curve, or Multiplier |
|---|---|---|---|
| Digital Integrated Circuits | CMOS | | |
| 101-500 Gates | 52 | 8 | 1.0 |
| 1001-2000 Gates | 70 | 8 | 1.0 |
| 10001-15000 Gates | 110 | 8 | 1.0 |
| Microprocessors | CMOS | | |
| 1001-2000 Gates | 50 | 8 | 1.0 |
| 10001-15000 Gates | 71 | 8 | 1.0 |
| Random Access Memory | CMOS, Static | | |
| 64Kbits | 170 | 8 | 1.0 |
| 256Kbits | 300 | 8 | 1.0 |
| Random Access Memory | NMOS, Dynamic | | |
| 64Kbits | 120 | 8 | 1.0 |
| 256Kbits | 180 | 8 | 1.0 |
| 1024Kbits | 270 | 8 | 1.0 |
| ROMS, PROMS, EPROMS | CMOS | | |
| 64Kbits | 55 | 10 | 1.0 |
| 256Kbits | 81 | 10 | 1.0 |
| 1024Kbits | 120 | 10 | 1.0 |
| 850nm Laser Diode | 15000 | 10 | 1.0 |
| 1550nm Laser Diode | 5000 | 10 | 1.0 |
| Discrete Resistor Fixed, Film | 2 | 3 | C |
| Discrete Capacitor Fixed, Al, Axial lead < 400µf | 30 | 7 | E |
| Relays, contactor | 560 | 3 | C |

**Table 3, Typical Device Failure Rates**

| | Integrated Circuits | | Discrete Semiconductor Devices | | All Other Devices |
|---|---|---|---|---|---|
| Quality Level | Hermetic | Non-hermetic | Hermetic | Plastic | |
| I, Some Device Quality Control | 1.5 | 1.8 | 1.5 | 1.8 | 1.5 |
| II, Average Device Quality Control | 1.0 | 1.0 | 1.0 | 1.0 | 1.0 |
| III, Tight Quality Control | 0.5 | 0.5 | 0.5 | 0.5 | 0.5 |

**Table 4, Quality Level Multiplier**

| | Electrical | Stress Curve |
|---|---|---|
| % Stress | C | E |
| 10 | 0.6 | 0.4 |
| 50 | 1.0 | 1.0 |
| 90 | 1.7 | 2.6 |

**Table 5, Electrical Stress Multiplier, $\pi_S$**

| Definition of Electrical Stress | |
|---|---|
| % Electrical Stress For Resistor | applied power/rated power |
| % Electrical Stress For Capacitor | Sum of applied dc voltage plus ac peak/ rated voltage |

**Table 6, Electrical Stress Definition**

| Operating Temp, °C | Temperature Stress Curve | | | |
|---|---|---|---|---|
| | **3** | **7** | **8** | **10** |
| 30 | 0.9 | 0.6 | 0.6 | 0.4 |
| 40 | 1.0 | 1.0 | 1.0 | 1.0 |
| 65 | 1.4 | 2.6 | 3.0 | 6.8 |

**Table 7, Temperature Stress Multiplier, $\pi_T$**

**Observations:**

As it can be inferred from the above formulas and tables, there are many factors that affect the reliability or potential lifetime of a device. Some devices are more sensitive to electrical stress, such as capacitors, resistors and diodes (not given). Laser diodes, ROMS, PROMS are very sensitive to high temperature operating environments. For example for these devices at 65°C, the expected lifetime will be almost 4 times worse than that operating at 40 °C. Quality control of incoming devices also has significant implications over their long term reliability. Activities such as burn-in or screening through temperature cycling, lot to lot control of components and periodic requalification are characteristic of Quality Level III (as given in Table 4), representing the tightest level of Quality Control. A product designed for high reliability performance would have all of these factors incorporated. The designer should be very cognizant of the steady-state operating temperature of the devices used in his design and should take measures to ensure that temperature sensitive devices are not heat stressed and discretes such as diodes, capacitors or resistors are not electrically stressed. Furthermore, an attempt should be made to minimize the number of high failure rate devices, and in general, keep component count to a minimum.

## 3.0 Hybrid Fiber Coax Availability Modeling

### *3.1 Model used for Telephony services:*

The average availability (up-time) goal for Plain Old Telephone Service (POTS) subscriber loop has been set to be 99.99% or no more than 53 minute average unavailable time per year. There are many reasons for this high level of reliability and availability, but one simple one is public reliance on telephone access for emergency 911 services. This objective, incorporating all network equipment between the local switch and the network interface (excluding the local switch and the customer premises terminal equipment), refers to the long-term average service to a typical customer.

Telephony-like services over HFC should therefore strive to achieve this goal if they are to be the only means of providing telephone service in the area. In our discussion herein, we will attempt to assess the challenges in meeting this goal and highlight some additional measures that need to be considered in order to achieve this goal.

According to the above definition, the "subscriber loop", consists of: Head-end Switch Interface Unit (HSIU), Head-end Power, Head-end Optics TX, Head-end Optics RX, Fiber-Node Optics RX, Fiber-Node Optics TX, Plant Power, Up to 3 cascade amplifiers and the Customer Interface Unit. (Figure 1)

Using (2), (4) and (5), the end-to-end availability measure of the above model can be expressed as follows:

$A = A_{HEP} * A_{HSIU} * A_{FOptics} * A_{ROptics} * A_{PP} * (A_{Amp})^m * A_{CIU}$ , where (8)

$A_{HEP}$ = Availability of the Head-End Powering equipment
$A_{HSIU}$ = Availability of the Head-end Switch Interface Unit
$A_{FOptics}$ = Availability of the forward optics
$A_{ROptics}$ = Availability of the reverse optics
$A_{PP}$ = Availability of the plant power
$A_{Amp}$ = Availability of Amplifier
m = Number of amplifiers in cascade, we will use m = 6, to account for reverse amplifiers too
$A_{CIU}$ = Availability of Customer Interface Unit

$A_{FOptics}$ and $A_{ROptics}$ are further broken down into,

$A_{FOptics} = A_{Flaser} * A_{Freceiver}$ , If no redundancy is used
$A_{ROptics} = A_{Rlaser} * A_{Rreceiver}$ , If no redundancy is used

If redundancy in the optics is used, then equation (5) can be applied.

In order to calculate the individual availability measures, knowledge of the MTBF and MTTR is necessary. The MTTR's were given in Table 1.

## *3.2 Analysis of the Model*

In order to get some idea about the expected availability measure of the end-to-end telephony model, we have to use some actual experienced numbers. In the above model, there is not much experience with the HSIU and CIU components, simply because these are products in the early phases of deployment.

To get an idea of the availability, we will create a hypothetical Customer Interface Unit. Here is the Bill Of Material for this hypothetical CIU (Table 8):

| Device Description | Quantity ** | Device FIT |
|---|---|---|
| 100k Gate Digital IC | 2 | 150 |
| Microprocessor | 2 | 100 |
| 256k DRAM | 2 | 180 |
| Relay | 4 | 560 |
| 256k PROM | 2 | 81 |
| **TOTAL CIU FIT** | | **3262** |

**Table 8, Hypothetical Customer Interface Unit Bill Of Material**

**- For purposes of illustration, we are oversimplifying the CIU construction. We are assuming that the quantities and types of devices in the above table are a most likely representative of the reliability of the real CIU.

Using the following coefficients and assumptions

$\pi_E$ = 1.5 (Outside Plant)
$\pi_Q$ = 1.0 (Quality level II, average)
$\pi_S$ = 1.0 (Designed, such that devices are not electrically stressed)
$\pi_T$ = 1.3 (All devices fall in temperature curve 8, assume 45 °C operating temperature)

$\lambda_{SSCIU}$ = 6361 FIT $\longrightarrow$ MTBF= 157,208 (hours)

$A_{CIU}$ = 0.9999618

One significant difference between the design of the CIU and the HSIU, is that there is great incentive in keeping the cost of the CIU as low as possible. This prohibits the use of redundancy in the design of the CIU. On the other hand, the HSIU can afford to employ equipment duplication and protection, since its cost is shared by many hundreds of potential subscribers.

To estimate $A_{HSIU}$ we assume that the FIT is ten times better than that of the CIU. This is not an unrealistic expectation, since the HSIU equipment will provide a much higher level of equipment protection, plus the operating environment will much more benign and stable. Therefore

$$A_{HSIU} = \frac{1572080}{1572080 + 4} = 0.99999745$$

To further continue the analysis we will consider 4 cases: (See Tables 9 -12)

Case I: Trunk and feeder with 20 trunk, 1 bridger and 2 line extender amplifiers. The annual return rate of the amplifiers are assumed to be 2%. For two-way telephony service the reverse path should be turned on. This results in an equivalent of 46 amplifiers in cascade in our reliability model. We see from Table 8 that the end-to-end availability is 99.93%. This number would further be aggravated if the dependence on commercial power were to be incorporated. It is common to see power outage in one segment of a cascade and not at the neighborhood. Taking all of these factors into account, it is easy to understand why so many people have a distaste for CATV service. Regardless of the power issue, we see that the Trunk and Feeder architecture is really not suitable for telephone services.

Case II: Hybrid Fiber Coax, no Optical Redundancy. Even though in this example the optics perform less reliably than the amplifiers, the end-to-end availability has improved to 99.97%

Case III: Same as Case II, with Optical Redundancy. Using redundant optical equipment, despite their relatively low reliability, improves the end-to-end availability to 99.987%

Case IV: Same as Case III, with more reliable amplifiers. Deploying 6 times more reliable amplifiers results in 99.995% end-to-end prediction.

Therefore, using the hypothetical CIU and HSIU, applying redundancy in the optics and assuming that
$A_{PP} = A_{HEP} \cong 1.0$, we were able to meet the Bellcore requirement of 99.99% long term availability. It is important to note that the only way that $A_{PP}$ and $A_{HEP}$ will be relatively high is through use of power back-up (i.e. batteries). Even then, the availability of emergency generators and emergency crews are essential in preventing extended power outages and in meeting the above requirement. The cable plant power has historically not been backed-up, therefore a cable plant being designed for telephone services must take all of the above into consideration

## 3.3 Other issues in consideration of availability:

- **Ingress:** The Cable plant reverse path suffers from a phenomenon called ingress. Ingress is the infiltration of 5-40 MHz energy off air (or through other means of energy coupling) from the various openings (primarily in the soft coax drop) into the plant. It is important to understand the effects of ingress on channel availability. All the considerations so far concentrated on electronic equipment reliability and up-time. Interference due to ingress can conceivably jam the upstream transmission, effectively making the service unavailable to the subscriber. Analysis of ingress and quantifying channel availability as a function of interferer characteristics and CIU characteristics is not the focus of our discussion . However, we will mention that how a vendors' product treats ingress can have significant bearing over the availability of the channel to the point that electronic equipment reliability could be overshadowed. For example, through the use of frequency agility, rugged modulation, and narrow carriers, a higher immunity to ingress can be expected. Given this, it is easy to compute $A_{channel}$ based on experimental information available on the plant ingress. Once $A_{channel}$ is computed, it can be treated as another serial link in the reliability block diagram analysis to provide the real service availability measure.

The ability to shut off or attenuate various parts of the return system will be critical to problem isolation and detection.

- **Proactive system tests**: This is simply finding faulty units before the customer does. An important feature that will differentiate various cable telephony products, is their background and foreground diagnostic capability. As an example, a CIU in concert with the HSIU and while the service is not being used, could be running background tests. If failures are encountered, the results are transmitted to a central location, whereby maintenance staff are dispatched. The fact that the CIU is primarily going to be located at the side of the house, makes maintenance activities quite non-intrusive. The maintenance crew can service the fault, long before the subscriber becomes aware of the problem. This is a major departure from the traditional entertainment cable and is one of the major advantages of the "demarcation point" concept. Bit-error-rate tests conducted in this manner at various frequencies can be utilized to "qualify" potential segments of return spectrum, before assignment.

- **Cables and Connectors**: These are other unknown sources that can and will cause service interruptions. Intermittent connections are a major source of ingress and signal degradation that can adversely affect channel availability. Intermittent connections are likely to result in very long outages since the trouble will most likely be in the house cabling. Access and troubleshooting become quite a problem.

-**Plant Power**: It was mentioned earlier that power reliability and availability must be taken into consideration and cannot be overlooked in providing telephone-like services. Batteries must in fact be in place and in good condition to be utilized under commercial outage situations. Status Monitoring may be well justified to verify this on a routine basis.

-**Software Reliability**: It will be critical in advanced systems to be able to diagnose problems before they occur, for preventive maintenance reasons. In the event of an unexpected failure, there is a high value attached to the ability of the product to automatically and correctly reconfigure redundant signal paths and raise appropriate notification. Since this intelligence is primarily realized through software its reliability becomes of paramount importance.

-**Return Spectrum Management**: There will be a high value placed on the ability to manage and control the various signals present on the return spectrum. Without this capability, many new and existing products and systems which utilize the return band for communication will overlap, contend and collide with high priority signals resulting in unpredictable communication and sub-optimal spectrum utilization.

-**Fiber Path Diversity**: Considering the extensive deployment of fiber, failures due to cable cuts could become a primary constraint. Optical cable path diversity may be well justified.

-**Return Power Sensitivity:** It has been shown that the return path can be overpowered if signals of arbitrary amplitude are permitted. The return amplifiers (just like forward amplifiers) will become non-linear if signal input power is too high. There is no effective gain control technique in the return path today and the return laser may be particularly susceptible to high power inputs.

## 4.0 Conclusion

This paper introduced the reader to some of the concepts of reliability analysis and some of the real world problems encountered in the Hybrid Fiber Coax plant. Quantifying service and channel availability to a subscriber is not a trivial task and requires a great deal of information gathering and analysis. Complex systems, in particular, where a great deal of module inter-dependency exists, cannot be easily analyzed. It was stated that electronic equipment up-time will not be the only factor in ensuring channel availability: issues such as connections and ingress must also be considered. We also concluded that optics redundancy and power back-up are very important in meeting telephone service availability requirements. It is possible to meet the availability requirement of 99.99% with proper network design, planning, and routine monitoring. Care must be taken in the selection of network components and their associated modulation and spectrum formats. OSS for these networks will be complex and necessary.

## 5.0 References


[1] "Reliability Prediction Procedure for Electronic Equipment" Bellcore TR-NWT-000332 Issue 3, September 1990

[2] "Methods and Procedures for System Reliability Analysis" Bellcore Special Report SR-TSY-001171 Issue 1, January 1989

[3] "Generic Requirements and Objectives for Fiber In the Loop Systems" Bellcore TR-NWT-000909, Issue 1, December 1991

| | Annual Return Rate% | Calculated MTBF (hours) | MTTR (hours) | Calculated Availability |
|---|---|---|---|---|
| Forward Laser | N/A | | 4 | |
| Forward Receiver | N/A | | 6 | |
| CoAxial Amplifiers | 2 | 438000 | 6 | 0.999986302 |
| 2-way Cascade Length | 46 | | | 0.999370066 |
| Reverse Laser | N/A | | 6 | |
| Reverse Receiver | N/A | | 4 | |
| CIU | 5.6 | 156429 | 6 | 0.999961645 |
| HSIU | 0.56 | 1564286 | 4 | 0.999997443 |
| **System** | | | | **0.99932918** |

**Table 9, Case I** **Trunk & Feeder 20 + 3 in cascade**

| | Annual Return Rate% | Calculated MTBF (hours) | MTTR (hours) | Calculated Availability |
|---|---|---|---|---|
| Forward Laser | 9 | 97333 | 4 | 0.999958906 |
| Forward Receiver | 9 | 97333 | 6 | 0.99993836 |
| CoAxial Amplifiers | 2 | 438000 | 6 | 0.999986302 |
| 2-way Cascade Length | 6 | | | 0.999917812 |
| Reverse Laser | 9 | 97333 | 6 | 0.99993836 |
| Reverse Receiver | 9 | 97333 | 4 | 0.999958906 |
| CIU | 5.6 | 156429 | 6 | 0.999961645 |
| HSIU | 0.56 | 1564286 | 4 | 0.999997443 |
| **System** | | | | **0.999671476** |

**Table 10, Case II** **HFC w/ no optical redundancy**

| | Annual Return Rate% | Calculated MTBF (hours) | MTTR (hours) | Calculated Availability |
|---|---|---|---|---|
| Forward Laser | 9 | 97333 | 4 | 0.999958906 |
| Forward Receiver | 9 | 97333 | 6 | 0.99993836 |
| CoAxial Amplifiers | 2 | 438000 | 6 | 0.999986302 |
| 2-way Cascade Length | 6 | | | 0.999917812 |
| Reverse Laser | 9 | 97333 | 6 | 0.99993836 |
| Reverse Receiver | 9 | 97333 | 4 | 0.999958906 |
| CIU | 5.6 | 156429 | 6 | 0.999961645 |
| HSIU | 0.56 | 1564286 | 4 | 0.999997443 |
| **System** | | | | **0.999876883** |

**Table 11, Case III** **HFC w/ optical redundancy**

| | Annual Return Rate% | Calculated MTBF (hours) | MTTR (hours) | Calculated Availability |
|---|---|---|---|---|
| Forward Laser | 9 | 97333 | 4 | 0.999958906 |
| Forward Receiver | 9 | 97333 | 6 | 0.99993836 |
| CoAxial Amplifiers | 0.3 | 2920000 | 6 | 0.999997945 |
| 2-way Cascade Length | 6 | | | 0.999987671 |
| Reverse Laser | 9 | 97333 | 6 | 0.99993836 |
| Reverse Receiver | 9 | 97333 | 4 | 0.999958906 |
| CIU | 5.6 | 156429 | 6 | 0.999961645 |
| HSIU | 0.56 | 1564286 | 4 | 0.999997443 |
| **System** | | | | **0.999946739** |

**Table 12, Case IV** **HFC w/ optical redundancy & improved reliability**

# CABLE POWERING INTO A DISTRIBUTED LOAD

**Doug Welch**
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## ABSTRACT

The process of modeling a distributed load cable system is performed. Basic power pack models are developed and tested. A cable network simulation is created using these models to study interactions between components of a distributed load. Steady state testing results show current waveform of a simulated power pack is highly dependent upon location in the network.

Input power disruption is used to study effect of power disturbance in distributed load network. Most significant result is that a small 8mS disruption causes a voltage drop at one node from 43.5VAC to 34.6VAC for a duration of over 50mS. Proves that small transfer times < 10mS can disturb network sufficient to exceed power pack hold-up times.

## INTRODUCTION

We felt it was important to more fully understand the characteristics of real cable system powering, which is basically a network of distributed loads. Since it is unrealistic to measure real cable systems (when amplifiers are thousands of feet apart) it is critical to develop accurate models to achieve any meaningful results. The first step in model development is defining the characteristics of a cable plant's components.

## INITIAL TESTING

### Test Setup for "Actual" Cable Plant

Our initial test setup consisted of five station amplifiers tied together with 875 and 500 cable (still on the spools), powered by a 4 amp 60VAC ferroresonant transformer (Figure 1). Since the cable sheath was uninsulated, we shorted out all of the sheaths with 12 gauge wire to avoid inconsistent measurements.

### Test Measurements of Cable Plant

In order to make current measurements we cut away the sheath just before and after each amplifier to expose the center conductor for our clamp-on current probe. The sheath cuts were jumpered to maintain sheath grounding integrity. The RF present on the output of the amplifiers made direct

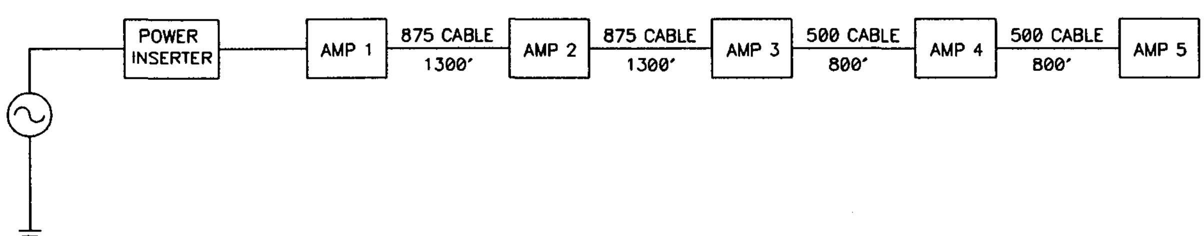


**Figure 1** - Initial Test Setup of "Actual" Cable Plant

measurements very difficult. Current was successfully measured using a clamp-on current transformer, which was not affected by the high frequencies. The voltages were measured at the standard test point inside each amplifier.

The clamp-on current transformer was modified with a precision resistor to produce a voltage waveform that accurately represented the current waveform at the test point. This allowed monitoring of the current waveforms throughout the cable plant test setup. These waveforms were recorded on a digital storage oscilloscope and photographed.

Data was gathered on the system's performance in both steady state and disruption conditions. The disruption involved different durations of outages occurring at different phase angles (relative to the incoming AC). The special test equipment used for this is discussed in detail later, along with the testing parameters.

The amplifiers were tested individually by removing them from their housings and plugging them, one at a time, into the first amplifier housing location. This confirmed that all of the amplifiers were behaving in the same way, and provided the data on individual amplifier characteristics.

## POWER PACK SIMULATORS

### Power Pack Simulation Model

Different methods of modeling the DC power packs found in typical station amplifiers were tried. The final design draws constant power, and is adjustable to match a specific amplifier configuration. A simplified design is shown in Figure 2. The op amp monitors the incoming AC voltage; when the voltage drops, the current draw through the transistor is increased to maintain a constant power draw from the AC line.

It was important to match the Power Pack Simulators (PPSs) performance as closely as possible to the characteristics of the power packs in the amplifiers we measured. The design is based upon a schematic of the input stage of the DC power pack, using the necessary component values to produce the proper current and voltage waveforms. The input impedence is closely matched (at 60Hz) by incorporating series inductance, a diode bridge, and a 1000$\mu$F capacitor. The overall performance is matched by the closed loop constant power sink. The current waveform from a typical station amplifier is matched by the waveform of our PPS, shown in Figure 3.

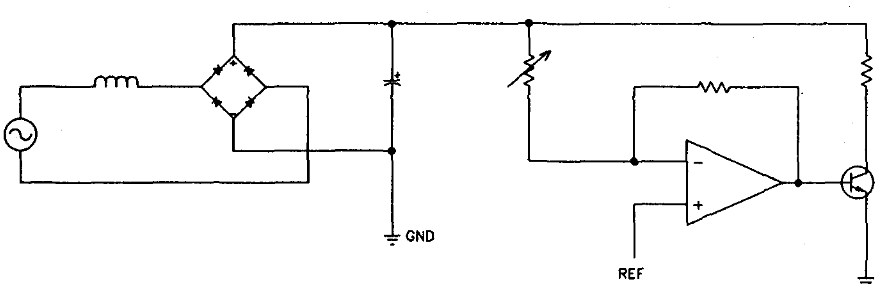


**Figure 2**
PPS Simplified Schematic

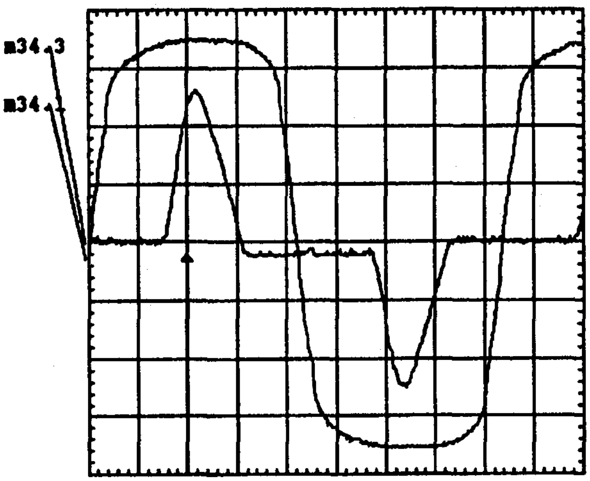

**Figure 3** - Single PPS Current Waveform

**Source Impedance**

A large number of PPSs were built and calibrated using a Variac on the output of a ferro supply with a 0.5Ω series resistor to measure current. Each PPS was tested and set for one of two power levels: 34.6W or 44.2W. The power levels are for two or three output station amplifers, respectively. Each PPS maintains a constant power level (within a few percent) over the range of 60VAC to 40VAC.

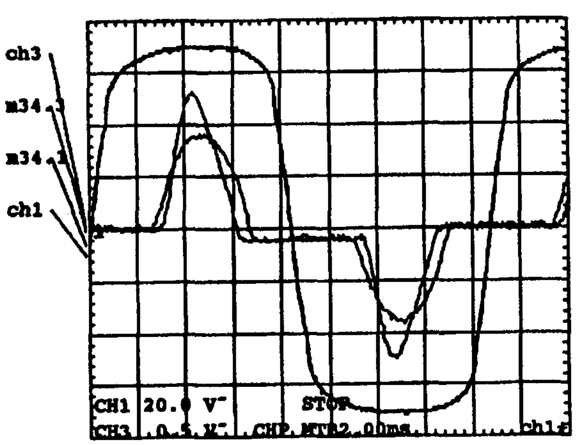


**Figure 4** - Source Impedance

The PPSs were then individually tested using different series resistors (for different coax cables) between them and the power supply. A Variac was used to maintain the input voltage to the PPS at 45.5VAC (rms). The current draw (and total power) was reduced when the resistance was increased, even with the same input voltage. The input current waveform changed significantly, as did the total power drawn by the PPS.

The waveform in Figure 4 shows the input voltage and overlays the two current waveforms created with an increase in source impedance from 0.5Ω to 2.6Ω. The current draw is spread out more over time due to the interaction of the series resistance and the large filter capacitor in the PPS. The capacitor cannot draw as much current as quickly with the series resistance, which reduces the current peak. The current draw also occurs earlier due to the fact that the voltage is actually slightly higher at the amplifier during the first part of the waveform. Even though the RMS voltage is the same, when the series resistance is significant, the voltage is initially higher at the PPS since there is no current being drawn at that time. Since the instantaneous voltage level is what causes the capacitor to start drawing current, this allows the capacitor to start drawing current at an earlier time in the waveform.

Another aspect is that the DC voltage across the capacitor is reduced slightly with increased source impedance. This could reduce the potential hold-up time in a power pack by reducing the energy storage level in the DC capacitor.

We believe this phenomenon is present in all switch mode power packs, since it is due to the electrical characteristics of significant source impedance. It is not clear what the measurable effect, if any, on hold up time might be. The more pronounced effect is the alteration of the current waveform. Of course this is of no consequence in a single amplifier, but when amplifiers are put into a distributed network, then it has an impact on the overall network current waveform, as discussed later.

## CABLE PLANT SIMULATION

### Network Configuration and Layout

The goal of the layout was to create a model that had some bearing on the real world, but without adding extra complexity which would mask the network response. The concessions to simplicity are powering from the end rather than the middle, and the equal lengths for all similar types of cable runs. The test network layout is shown in Figure 5.

The components being modeled are: two main cable runs of 875 cable with all other cable runs being 500 cable. The total loop resistance for each run of 2000 feet of 875 cable is 1.1Ω, and the total for 1500 feet of 500 cable is 2.58Ω. All of the amplifiers are of the "station" type, with full reverse path capability, using the same switch mode DC power pack. This approach defines the power requirements according to the number of outputs: a two output unit requires 34.6W and the three output requires 44.2W.

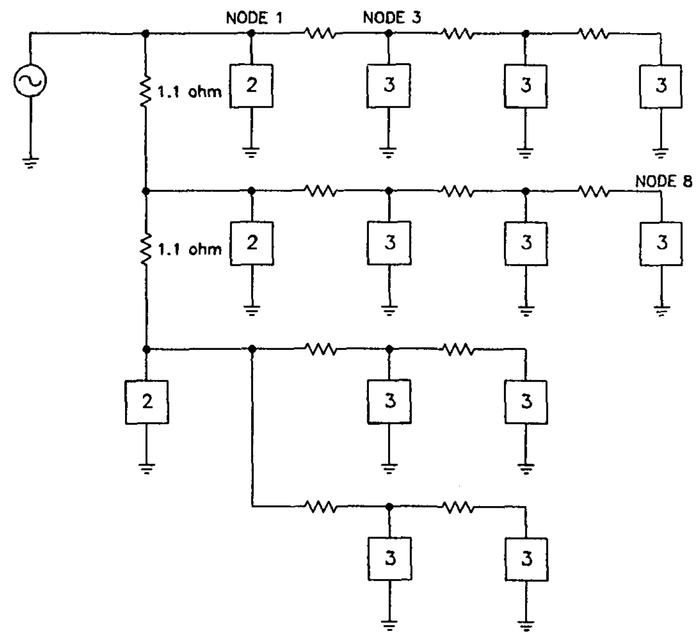


**Figure 6** - Test Network Schematic

An electrical engineer looks at this cable network in a different way. Figure 6 shows the same network in a traditional schematic format, which we will use for the rest of this paper. It is assumed that there is no significant impedance (at 60Hz) through the amplifier so the input and output are considered to be the same node. Each PPS (amplifier) is shown as a load interconnected by the resistors which match the loop resistance. Another simplification is that the loop resistance is treated as a single resistance in the power line. All of the amplifiers have the same return potential (ground) to enable accurate voltage and current measurements.

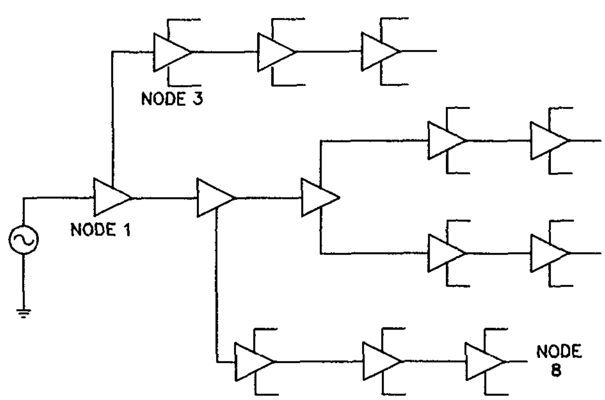


**Figure 5** - Test Network Layout

## NETWORK WITH STEADY STATE POWERING

The first group of scope displays show the typical waveforms throughout the network during continuous steady state powering. There are three signals on each display; channel 1 is current into the PPS at that node, channel 2 is the voltage at the PPS node, and channel 3 is the voltage at the power supply that drives the network.

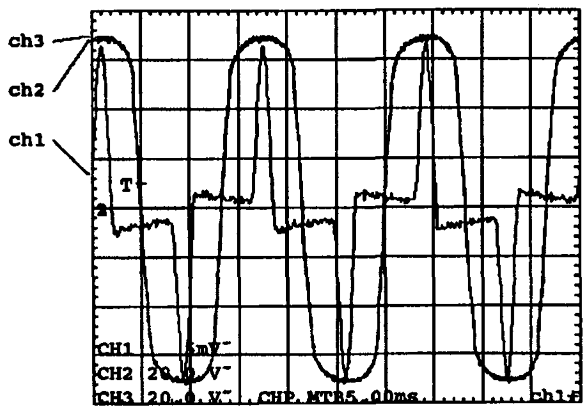


**Figure 7** - Node 1, Steady State

Rather than go through all of the data, we will review some of the basic characteristics. Figure 7 shows Node 1. Of course since this node is at the power supply the two voltage waveforms are identical. Note that the current waveform is the same as a single amplfier by itself, shown earlier. The current peaks at 1.7 amps with a rise time of 1mS for a duration of about 3mS. This is characteristic of a capacitive load because the current draw occurs only when the input voltage exceeds the voltage across the capacitor. The fact that the current has a long rise time of 1mS is due to the series inductance in the PPS. The input voltage waveform is quite rounded on top so the highest voltage is in the middle of the waveform, which is when the current is drawn.

The next node, 3, already shows the effect of cable resistance. This is an isolated branch of the network with three PPSs and three cable runs. One of the obvious factors in cable plant interactions is the effect of total current for a branch creating a large voltage drop over the first run of cable. This affects the voltage waveform seen at node 3, as shown by Figure 8. Rather than having the rounded top seen at the power supply, the voltage waveform is much flatter, with a lower peak value (61V). This has the predictable effect of reducing the voltage on the DC capacitor (from 67.0VDC at Node 1 to 59.6VDC) and changing the current waveform. The current draw starts sooner due to the peak input voltage occurring sooner. Other effects include a reduction in peak current, down to 1.3A, and an increase in duration of current being drawn. Note that the rise time remains about the same in this case.

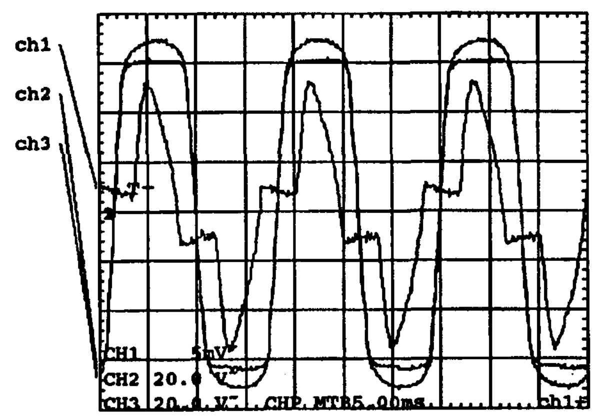


**Figure 8** - Node 3, Steady State

The most pronounced effect of cable plant powering is typically seen at the far reaches of a network, where the common condition is being close to "voltage starvation". This is illustrated at three locations in this test network, with the worst being node 8. The voltage at this point is 43.5VAC (rms). In Figure 9 this is seen as a very flat top voltage waveform, with a peak voltage of about 47V.

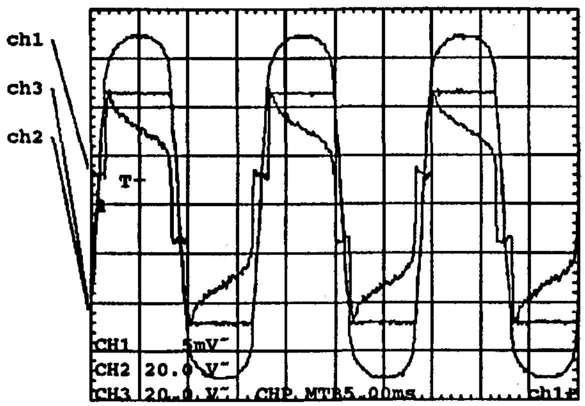


**Figure 9** - Node 8, Steady State

The current waveform seen here is very informative. Since the voltage peak occurs very early, the current reaches its peak (of 1.1A) about 1mS after zero crossing (and the rise time is also reduced to about 500$\mu$S). The duration of current draw is 7mS, which is the entire time the voltage waveform remains high. It is also interesting that even though the current drops over time, it is still drawing more than 500mA when the voltage turns off.

All of these effects are combined in Figure 10. This display shows the voltage from the power supply and the current going into the test network. In the steady state condition the current draw is concentrated mostly in the center of the voltage wave-

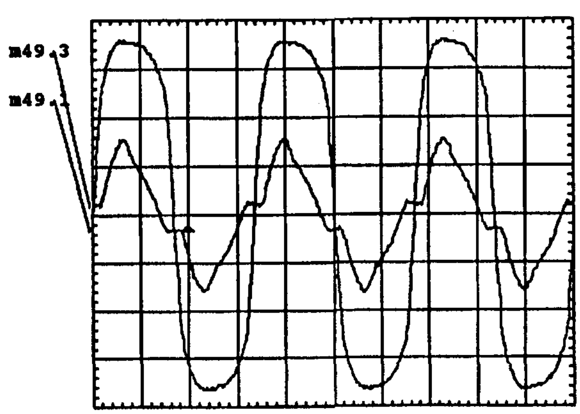


**Figure 10** - Network, Steady State

form, but it is also spread out for most of the waveform. It is obvious that there would be a loss of energy to some part of the cable system if the voltage were disrupted almost anywhere in its waveform.

## POWER DISRUPTION

### Disruption Analysis

There are many different views on transfer time including how it is measured and its degree of importance. An 8mS transfer time has traditionally been viewed as acceptable for Standby power supplies. This is typically justified by pointing to the fact that most amplifiers have a hold-up time on the order of 20-40mS. Even some recent specifications for "UPS" powering of HFC systems only require a 5mS transfer time. This indicates that those writing such specifications feel there is a considerable safety margin in such a time.

It is not our intention to try to mimic how others (ourselves included) perform a transition to standby operation. It is only to introduce a very mild but highly defined disruption into a cable network and analyze the results. In order to generate the most meaningful data in any experiment, it is necessary to eliminate unnecessary complications and variables, and to have a consistent, repeatable, and verifiable test setup. Our goal was to find a way to measure the response of our test network to some type of powering disruption, which would be in some ways relevant to what happens during a non-zero transfer time.

We felt an 8mS disruption (½ of an AC cycle) is actually quite easy on the test network because of this approach: The interruption is not in the output of the power supply to the cable system, as would occur with a typical transfer relay. In this setup the ferro is still capable of supplying energy to the test network during the disruption, which it does. There is no inverter which eliminates complications of AC detection, response time, and performance characteristics. The same ferro comes back on to supply the test network after the disruption, with the full power of the utility grid as its source.

## Special Test Equipment

We developed a special piece of test equipment to allow a disruption of the incoming AC power that feeds our power supply which in turns feeds our test network. The Lectro Controlled Disruption Device (CDD) is a powerful research tool. It monitors the incoming AC voltage waveform and disrupts (turns off) the AC power at a specified phase angle for a specified duration. Phase angle is a way of measuring a periodic sinusoidal signal. The distance from zero crossing to zero crossing (one half of an AC sine wave) is defined as 180 degrees, which takes 8.33mS at 60Hz.

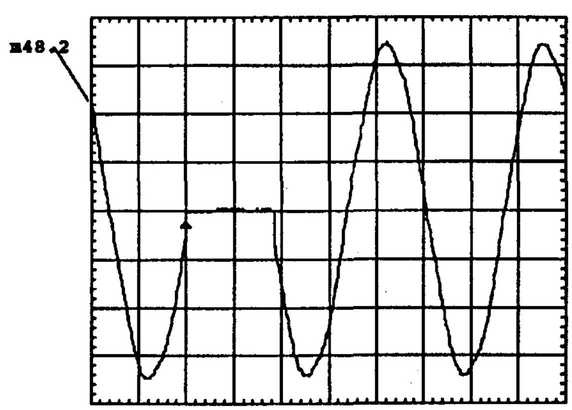

**Figure 11** - 0° Disruption, 8mS

The CDD can be adjusted in one degree increments from 0 to ±180 degrees. This allows for very fine adjustment of when

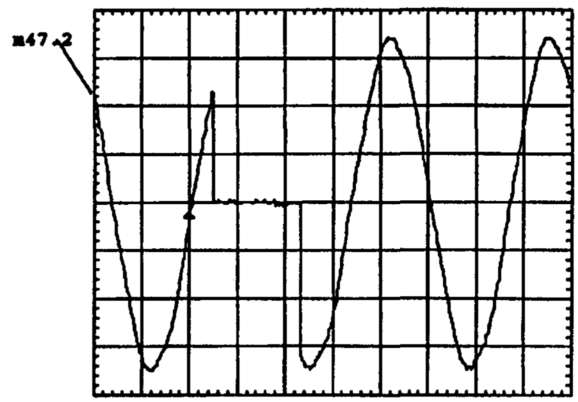

**Figure 12** - 45° Disruption, 8mS

the AC signal is removed from the device under test, in this case our power supply. The duration of the outage is also programmable in 1mS increments from 1mS up to 16mS (in its present configuration). This means we can consistantly generate a controlled disruption of the power line in order to see its effect on our test network.

Figures 11, 12, and 13 show the output of the CDD in the three different test conditions. All of these disruptions are for a duration of 8mS; the difference is in the phase angle. The first is at 0 degrees, the second is at 45 degrees, and the third is at 90 degrees.

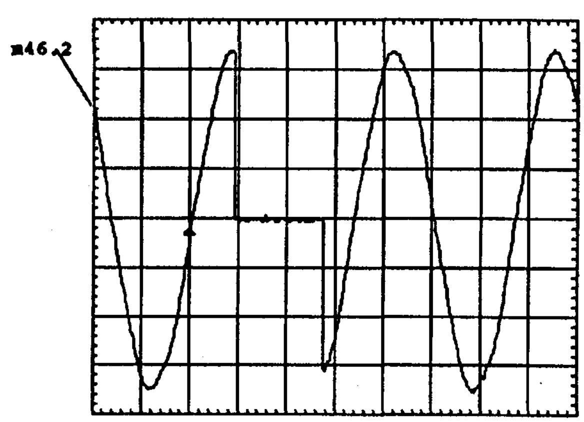

**Figure 13** - 90° Disruption, 8mS

## NETWORK WITH POWERING DISRUPTION

### Test Parameters and Setup

Three phase angles: 0°, 45°, and 90° were used as the test parameters for the disruption testing. This defines when the disruption starts relative to the incoming utility AC voltage waveform. All of the disruptions were set for a duration of 8mS, as described earlier. As a reminder, the 0° disruption cuts off a full half cycle, the 90° disruption shuts off from one positive peak to the next negative peak, and the 45° disruption cuts off ¾ of one half cycle and the first ¼ of the next.

The displays consist of channel 1 as the current waveform at the node under test, channel 2 as the output voltage from the power supply, and channel 3 as the voltage at the node under test. All of the waveforms are AC and centered on the display. A small triangle, located two divisions in from the left, indicates the trigger signal input from our CDD. Note that this trigger fires on the last zero crossing prior to the disruption (it is simultaneous with the disruption at 0°).

### General Observations

There are some common traits seen in all of the following displays, Figures 14 through 22. Several are inherent to a ferro-resonant transformer, while others relate to the PPS and the network itself.

First is the fact that even with a positive power disruption, there is still voltage being supplied to the load by the ferro. As shown by the displays in Figure 14, there is also some current being supplied at this time. Note that the scale for current in this display is 10A per division. Power is being delivered to the load with no power coming in. This is a clear example of the energy storage capabilities of a ferro transformer. Of course there is a price to pay for this "free" energy, as discussed below.

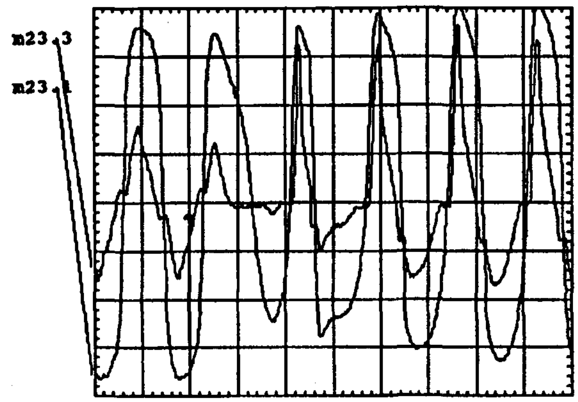

**Figure 14** - Network, 90° Disruption

The price is that the ferro must recover its energy when the input power is restored in order to become a stable regulated voltage source. One impact is a voltage overshoot during the first half cycle after return of power from a disruption. In this case this is a big benefit to the test network because a voltage spike provides assistance to all of the PPSs in recovering the energy they lost during the disruption (note several current spikes over 30A in Figure 14). This voltage overshoot is common in closed loop control systems (which is what the secondary winding of a ferro is). Along with the spike there is a frequency distortion during the first cycle, which is another ferro characteristic during energy loss. It is important to remember that these ferro actions are typical of a stand-alone ferro under these disruption conditions; they do not occur in a properly designed zero transfer time power supply.

Another common trait of the test network is the ability of the PPS to store excess energy. This is seen during the ferro overshoot when the current spikes throughout the test network. Some of the PPSs can go several cycles before needing to draw current again.

The last general observation is the confirmation that interactions exist in the test network. This is seen in the different voltage and current responses at the different nodes of the network during a specific disturbance. It is also evident from the significantly different reactions that the same nodes have to each different disruption. The cause and effect relationships are highly interdependent, making predictions difficult. As one node within the network does not pull current because it stored enough on the previous cycle, it means the voltage drop along its feeder cable is reduced by that amount, which potentially raises the input voltage for the following nodes. However, if a node pulls more current because its input voltage was raised, it drops the voltage for the following nodes. All of these interactions are happening at the same time, even within a particular half cycle. That is why detailed measurements of a test network under known conditions provide the best insight into this interaction.

## Current Waveform Analysis, 0° Phase

At node 1, both power supply and input voltages are the same since there is no series resistance element (Figure 15). The power supply voltage drops in the first half cycle (during the outage) from its normal peak of 69V to a peak of 54V. The next half cycle peak is off scale (above 80V peak) since the power is now restored to the ferro. The ferro is still recovering during the next half cycle with a peak of 64V, with some distortion.

The current waveform during the first

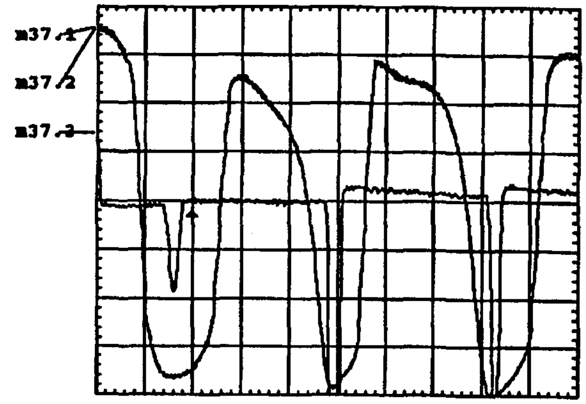

**Figure 15** - Node 1, 0°

half cycle (when the voltage drops) shows no current draw, followed by a sharp current spike (along with the voltage spike) during the next half cycle. After this point (not shown in figure) the PPS does not draw any current for over 60mS (7½ cycles) even though the node voltage is clearly back to normal within two cycles.

At node 3 (Figure 16) the input voltage during the outage cycle is identical to the power supply voltage. This is because virtually none of the PPS devices draw any current during this time so there is no voltage drop, even out to the end of this cable run. The next half cycle (as power is re-

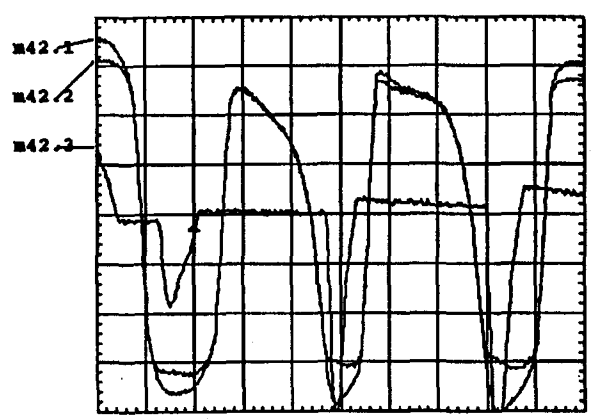

**Figure 16** - Node 3, 0°

stored) shows that while the power supply voltage peaked at over 80V, the input voltage at this node only peaked at 69V, which is not much above its nominal peak of 64V. As many of the PPS devices close to the power supply benefit from the voltage by pulling more current, the price is a significant increase in the voltage drops across the network cables. The next half cycle shows no current draw, followed by a slightly larger than normal draw for the next. This erratic pattern repeats for several cycles before damping back to its steady state condition.

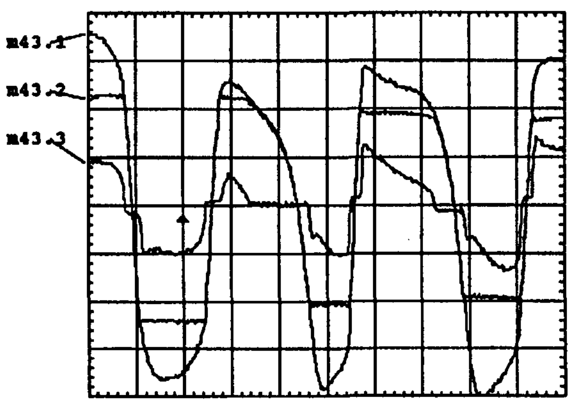

**Figure 17** - Node 8, 0°

Node 8 (Figure 17) displays a completely different current response than the previous two nodes, primarily due to its location (see Figure 6). First is the fact that it draws current during the first 2mS of the outage. Even though the ferro's voltage is dropping, the lack of currrent draw else where in the network allows node 8's input voltage to be almost normal for a few mS. The next half cycle (when power is restored) is very telling: even though the ferro's voltage goes above 80V, the input voltage at 8 drops to less than 42V. Current is still being drawn, though only during the last few mS of the cycle, as its input voltage slightly increases. The input voltage is still dropping (around 38V peak) during the third half cycle while current spikes to over 1.5 amps. Unlike nodes A1 and A3, which pulled peak currents for one or two half cycles then drew very little, node A8 is pulling high peak currents for several cycles after the disruption and never draws less than its nominal peak.

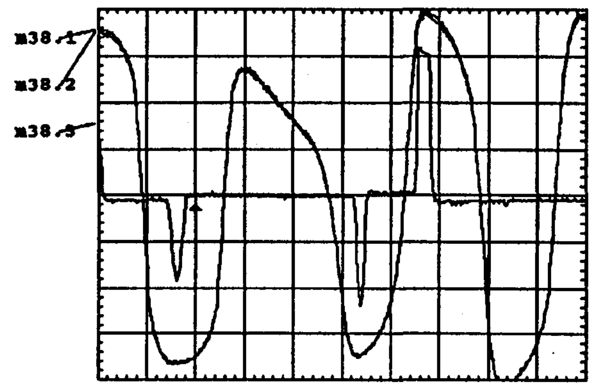

**Figure 18** - Node 1, 45°

**Current Waveform Analysis, 45° Phase**

The differences between the nodes 1, 3, and 8 were discussed above, so only those conditions that are different due to the phase angle will be covered. Node 1 (Figure 18) shows a very similar response, except under this condition the ferro generates two voltage spikes over 80V peak instead of one. This leads to two current spikes, then the typical

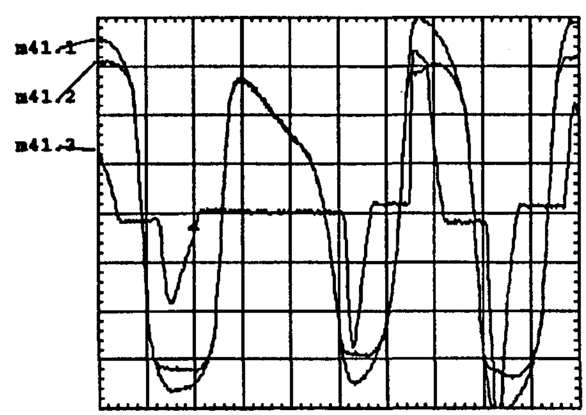

**Figure 19** - Node 3, 45°

multicycle duration of no current. Node 3 (Figure 19) shows a similar pattern as before with the modification of the second voltage spike.

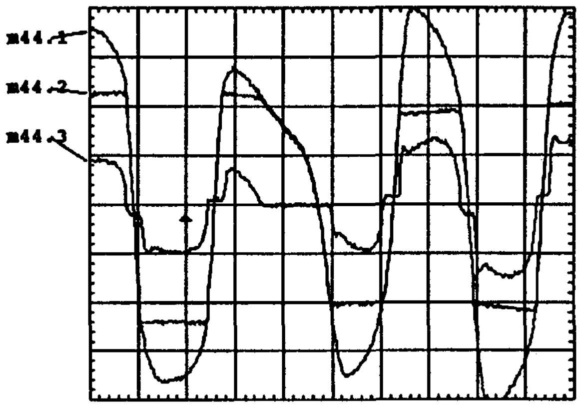

**Figure 20** - Node 8, 45°

Node 8 (Figure 20) again pulls current during every half cycle, but this time it recovers back to steady state in about three cycles. It is interesting that node 8 recovers quicker at 45° than at 0° or 90°. This is probably due to the interactions of all of the plant simulation components finding a balance under this set of conditions.

**Current Waveform Analysis, 90° Phase**

The node 1 display (Figure 21) shows the impact of this disruption on the power

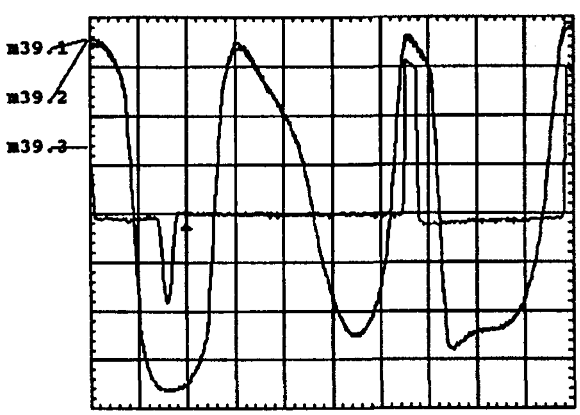

**Figure 21** - Node 1, 90°

supply waveform. It reaches its normal peak voltage, then drops off quickly. The next

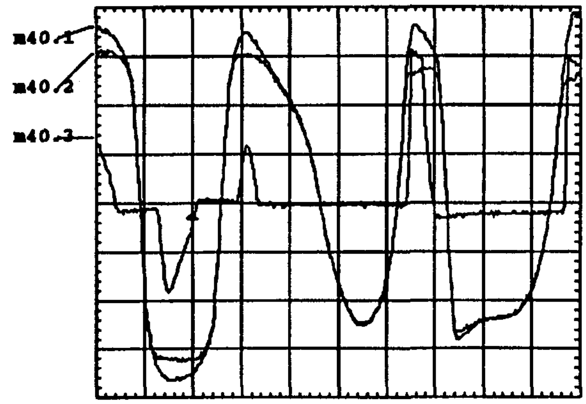

**Figure 22** - Node 3, 90°

half cycle is much lower with only a 46V peak. Energy returns in the third half cycle with the voltage peaking around 80V. Node 1 shows two complete half cycles of no current, followed by a current pulse corresponding to the return voltage pulse.

Node 3 (Figure 22) demonstrates how its lower nominal voltage masks the effect of the power supply's voltage drop as it pulls a significant portion of its normal amount of

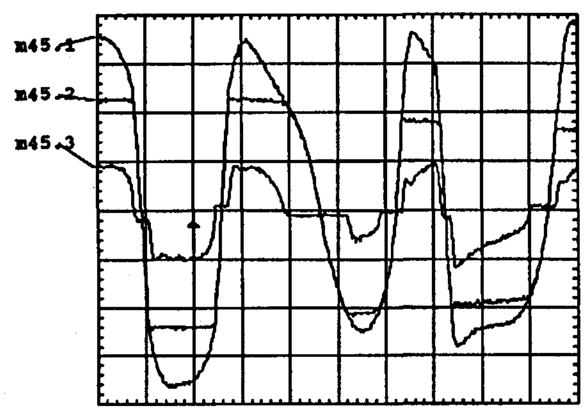

**Figure 23** - Node 8, 90°

current. Again the effect of current flow throughout the network is seen in the differences (or lack thereof) between the input voltage and the voltage at the input of 3.

Node 8 (Figure 23) pulls almost all of its current during the first half cycle, and even manages a small amount in the second half cycle. This is followed by many different patterns of current waveforms. Node 8 is attempting to pull current from anywhere it can.

### Voltage Waveform Analysis, 90° Phase

It turns out that the most informative view of the effect of an 8mS disruption at 90° is the voltage waveform. Figure 24 shows the input voltage at node 8. This

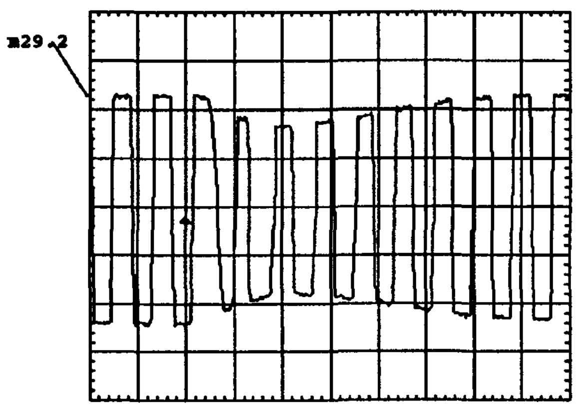

**Figure 24** - Voltage Drop, Node 8, 90°

clearly shows a drop in voltage over several full cycles. More specifically, it shows a drop in peak (nominal) voltage from 47V to a minimum of 35V. What is surprising is that it also shows a drop from 47V to below 37.5V for more than 50mS. This can be related to an amplifier normally running at 43.5V rms that goes down to 34.6V rms for at least 50mS, all from a single, simple disturbance of only 8mS.

## ZTT POWERING OF NETWORK WITH DISTURBANCE

A Lectro ZTT was installed to power the test network. The CDD connected to the AC input of the ZTT to disrupt power as in the previous test setup. The resultant waveforms, shown in Figure 25, are for node 8 with an 8mS disruption at 90°. The voltage at the power supply shows a slight drop, but the voltage at node 8 remains constant, except for the slight rise in peak voltage (2V) during the second half cycle. There is no voltage loss at node 8 with the ZTT, as was previously shown in Figure 24.

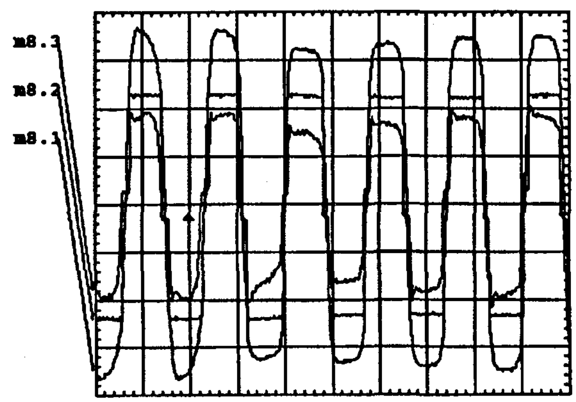

**Figure 25** - ZTT Power, Node 8, 90°

The current at node 8 shows a slight rise in the second half cycle which matches with the voltage rise. For the next few cycles the current drops slightly before returning to normal. This waveform is nearly identical to the steady state waveform seen in Figure 9 (except for different time base).

## RESULTS

This is by no means an exhaustive review of the impact of disruptions, resulting disturbances, transfer times, etc. It is only the tip of the iceburg in developing a solid understanding of distributed load powering.

The most significant result of this limited testing program is the significant voltage drop for such a long period of time from such a short disruption. In this setup, the worst condition found (so far) is at node 8 with a disruption of 112°.

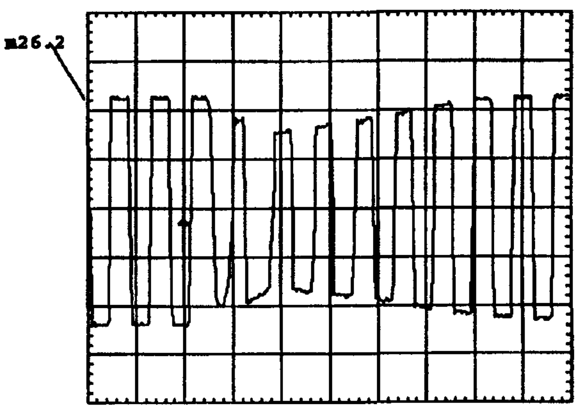

**Figure 26** - Node 8: 112°, 8mS Disruption
Voltage drop: 43.5VAC to 30.1VAC
Duration: 69.6mS

The depth and duration of the voltage drop at node 8 is due to the characteristic interaction seen in a distributed load. The series resistances that represent the cables of a real system are the elements that modify the voltages throughout the cable plant during any disturbance in powering, no matter what the source.

## CONCLUSIONS

It is obvious that transfer time, or anything that disturbs the power source output, has significant effects. It is clear that trying to claim transfer time is acceptable because it is smaller than an amplifier's hold-up time is wrong; there is no direct relationship.

Some large MSOs are starting to require low voltage limits on amplifiers, where the DC power pack goes off line if the AC input voltage drops. It is realistic to believe that an amplifier with this feature would probably shut itself down and lose its output if located at node 8 in this test network.

Considering the direction of the industry towards higher reliability and the desire to provide lifeline telephony services, the powering of any cable plant should include provisions to eliminate any source of power disturbances.

## ACKNOWLEDGEMENTS

I want to thank Larry Ross for his help in setting up the initial cable plant test, and for his input in developing the Test Network layout.

I also want to thank the Engineering staff at Lectro Products for their many hours of effort in testing and data analysis that made this paper possible.

# CHALLENGES AND SOLUTIONS IN THE INTRODUCTION OF DIGITAL SERVICES IN HOME COAX WIRING

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*Abstract*

*It is well recognized that the home coax wiring is a most variable and challenging element of a CATV network, particularly when handling digital signals. This paper identifies a wider than previously described range of issues and how these issues may be effectively dealt with without the need to rewire the home. Variable or intermittent home wiring performance, caused by subscriber changes in equipment configuration, is an additional challenge. A solution set to these and other issues is proposed which keeps the cable company in control, provides for maintenance, and permits existing and variable home coax distribution to be used for digital services without additional truck-rolls. The paper concludes with some results of lab measurements and plans for field transmission robustness verification.*

## CHALLENGES

### Upstream Transmission

Transmission of upstream commands or data from set-tops or future home data modems creates a very significant challenge to the cable industry. A very high signal level is required from a set-top converter in order to deliver (via the home wiring, splitters, drop and tap) an adequate level to, for example, a line extender return amplifier. This signal level is likely to interfere with TV set video, particularly if upstream frequencies in the 30 MHz to 40 MHz band are used. In addition, any coax connection looseness or poor shielding may cause unacceptable radiation or signal egress. Furthermore, any unintentional (or intentional) continuous upstream transmission from a set-top in one home could easily disable upstream signals from many or all other homes served from the same node. This would leave the cable company with a very difficult maintenance issue.

### Downstream Transmission

Downstream analog transmission impairments create gradual TV picture quality reduction. In contrast, digital transmission works exceedingly well up to a given degree of impairment and then totally fails beyond this threshold. Use of Forward Error Correction (FEC) techniques, normally required in the cable feeder plant, extends the available interference susceptibility margin in the home wiring by 5 dB to 10 dB, however the threshold of non-performance becomes steeper. A solution is required which overcomes potential problems of interference into the downstream path of home coax wiring.

Sources of interference include conducted emissions from "cable ready" TV sets, ingress from local broadcast stations and hand-held cellular equipment, domestic electrical appliances, etc. The solution must also work effectively in home coax wiring which has been installed or changed by the consumer. Even if the local cable company were to re-wire the home and install improved performance splitters, subsequent changes in TV set, set-top or VCR configuration by the consumer may still result in varying digital transmission performance requiring ongoing cable company action. A more radical approach is needed to ensure digital performance and flexibility without the need for ongoing cable company maintenance of home coax distribution.

### Spectral Efficiency

While trying to resolve digital downstream issues in the home wiring there is a need to retain or improve upon spectral efficiency in the feeder plant. This is required initially to defer the cost of feeder plant upgrades requiring amplifier respacing. However there is also a strong need to provide unlimited service variety to compete in the TV services marketplace.

## TECHNICAL SOLUTIONS

### Upstream Transmission

A proposed total solution to the set of upstream issues requires that the upstream signals from each home or MDU be gated or regenerated before being transmitted up the drop.

A small network interface module (NIM), located at the entry point of each residence (or MDU) and preferably locally powered, rapidly polls each device within the home(s) and accumulates messages or data. At fast but somewhat less frequent intervals the headend communicates with each NIM and receives the contents of its buffer. If the NIM buffer was previously filled to capacity this enables it to again poll home devices. Since the headend determines the order and priority of transmission from each NIM, it can assign, dynamically, individual upstream bandwidth from each home or MDU This is important in a multiple application services architecture.

The headend can also control each NIM in order to determine its frequency of polling home devices. Thus the headend also has the ability to deny upstream transmission from specific device within any home or MDU. Homes not subscribing to digital services should be fitted with 5 MHz - 40 MHz traps to prevent their interference with the new services. If required a bypass may be provided to accommodate specific existing RF IPPV devices.

A wide range of advantages accrue from the above upstream system design :

- since the upstream signals need only pass from the home devices to the NIM, much lower signal levels are needed - typically 30 dB lower than those needed to also overcome the drop, tap and feeder losses. This lower signal level has potential to reduce radically the home wiring leakage (egress) impact.

- since home wiring upstream signals have exclusive use of local spectrum, very robust, low cost, binary modulation techniques may be employed. In effect spectrum is reused in each home or MDU. Moving the return channel spectrum in home wiring to a much higher frequency, where interference is less prevalent, avoids the need for spectrum management in the home wiring, thus simplifying set-tops or other home devices.

- upstream transmission performance from the NIM to line extender reverse amplifiers is relatively noise-free (no home wiring ingress). Thus a <u>greater proportion of the upstream spectrum is usable</u> . Also higher order modulation methods may be used - QPSK universally today, probably 16-QAM in modernized plant. Thus the <u>overall upstream capacity of cable feeder systems can be increased significantly</u>. This creates the opportunity to defer the need to use top-of-band return channels in the feeder, thus avoiding the need to change out the line extenders.

- dynamic management of feeder upstream spectrum, to avoid dynamic sources of interference, becomes simpler since this does not involve controlling the frequencies of set-tops or other home devices. Thus there is some potential to reduce set-top complexity and cost.

- since upstream signals from each home or MDU are only transmitted on the cable feeder from the NIM, consumer equipment cannot intentionally or accidentally interfere with the upstream service of others. Furthermore, this means that any home having excessive electrical interference sources, such as hairdryers, electric tools, etc., will not interfere with the upstream transmission of other homes.

- since the headend provides control, the upstream bandwidth from each home, and even individual devices within a home, can be varied according to service needs.

- since the upstream spectrum from home wiring is filtered out and replaced by buffered digital signals from NIMs which transmit one-at-a-time, aggregation of interfering sources common to a neighborhood, such as Ham or broadcast transmitters, is much less likely.

## Downstream Transmission

Downstream transmission in cable feeder and in-home coax wiring have mutually conflicting requirements. In the feeder where transmission performance is good there is the need to optimize for spectral efficiency with potential deferral of re-engineering the coax plant until traffic needs expand.

In home wiring where interference and disrupting conditions generally exist there is the need to provide very robust transmission. However spectrum is abundant if only the channels being viewed exist on the home wiring. Additionally, if a single digital band containing fewer, time multiplexed, channels is transmitted, its level may be higher than when the total available spectrum carries many broadcast digital bands.

For the feeder, the cable industry is relatively confident that 256-QAM or 16-VSB digital modulation may be used, offering between 6 and 7 bits per second for each Hz of bandwidth (b/s/Hz). However the maximum level of each 6 MHz digital band needs be kept to about 8 dB below that of AM VSB vision carrier levels to avoid intermodulation with existing services.

By the time a 750 MHz band digital signal reaches the home it will have been attenuated to a level of around 600 µV. In average home wiring this level will likely be of the order of only 300 µV at a digital set-top. For 64-QAM a signal to interfering carrier ratio of at least 25 dB (under typical conditions) is required to achieve acceptable Bit Error Rates (BER). Thus the maximum permitted level of an interfering signal introduced into the home wiring near to the digital set-top is only approximately 16 µV (-36 dBmV). However, measurements of samples of "cable ready" TV sets indicate "antenna terminal", in digital band, conducted emissions in the hundreds of microvolts, in some cases in excess of 1 mV (0 dBmV).

Using the above numbers, there would seem to be a strong case to tune and demodulate a QAM (or VSB) band at a NIM and to re-transmit the acquired digital signal in a more robust, low-cost, less spectrally compact, manner in the home. This has a number of advantages and trade-offs :

- provides a solution to conducted or radiated interfering signals from TV sets, broadcast transmissions in same band as digital cable services, etc.

- offers robustness against electrical interfering sources such as hairdryers, vacuum cleaners, etc., particularly when connectors are not fully tightened or coax is improperly crimped, or where shielding braid is insufficient or damaged.

- a single, simply modulated, digital band in the home wiring allows much higher carrier signal level but since energy is spread across a greater bandwidth this level meets with accepted practices and standards.

- much lower cost digital set-top boxes as no tuner, echo-canceling or error correcting circuits are required (Tuners for high order modulation digital signals are much more costly since very low phase noise performance is required). Only a single tuner, QAM demodulator and error correction circuits are required (in the NIM) per home or MDU. Thus when more than one digital set-top per NIM exists or the QAM receiver provides any additional non-video service, overall system cost is lower than in conventional digital cable architectures. The small cost of providing service selection and modulation at the headend becomes absorbed by reduced network upgrade costs and, at the same time, variety of services is unlimited - a definite advantage in competitive markets.

- the cost of housing and powering the QAM and home wiring modem circuits can be shared with the need to have a NIM for the upstream transmission. Where two or more digital video services per NIM exist (multiple set tops / home or MDU situations) or as soon as other services are considered, the overall economics and flexibility of the proposed architecture has advantages over conventional, piecemeal, approaches. Examples of added services include data, energy management or second telephone service. A combination of voice, data and two-way video could provide an excellent multimedia work-at-home capability. The architecture described is one which will enable cable companies to compete effectively as broadband telecommunications access carriers of choice, primarily for residential customers but also for medium or small businesses.

- transmission of only requested services on each 6 MHz digital band, serving perhaps 8 to 12 homes each, reduces the opportunity for piracy.

Choice of in-home Digital Modulation

The simplest, most robust and lowest cost digital modulation is generally of binary form. Phase, rather than amplitude, modulation is generally more tolerant of interfering signals and non-linearity. Differential Binary Phase Shift Keyed (DBPSK) modulation allows a choice of very simple non-coherent or simple coherent receiver techniques. While micro-reflection cancellation is possible, the robustness of DBPSK techniques make this non-essential. Thus the choice of DBPSK modulation, using today's silicon technology, offers lowest cost as well as good potential for future demodulator technology evolution.

The choice of bit rate is dependent on service requirements. A bit rate which exceeds that of 256-QAM (or 16-VSB) in a 6 MHz band (approx. 38 Mb/s, allowing for FEC) simplifies both NIM and system design.

An in home wiring bit rate of around 52 Mb/s provides sufficient reserve capacity for passing control or polling signals from the NIM to the set-top and for a traffic and control return path using Time Compression Multiplexing (TCM), or "ping-pong", transmission techniques. TCM also offers lower filter costs and provides for future upstream/downstream bit-rate flexibility

The choice of spectrum for the 52 Mb/s DBPSK TCM signal is governed by available spectrum in the home wiring. A center frequency of around 930 MHz avoids the cellular telephony transmission spectrum and passes, albeit with acceptable additional attenuation, through existing domestic coax cable and splitters. Although this frequency is relatively high, the robustness of DBPSK allows for transmission in plant unsuitable for other modulation formats. PSK demodulators are, like those of FM, particularly insensitive to level variations and signal distortions.

## OVERALL ARCHITECTURE

A summary of the proposed architecture is illustrated in Figures 1 to 4 :

The proposed Cable System Architecture is shown in outline form in Figure 1. At the headend, digital services received from fiber or satellite facilities, or stored locally, are selected for each 6 MHz band of modulation.

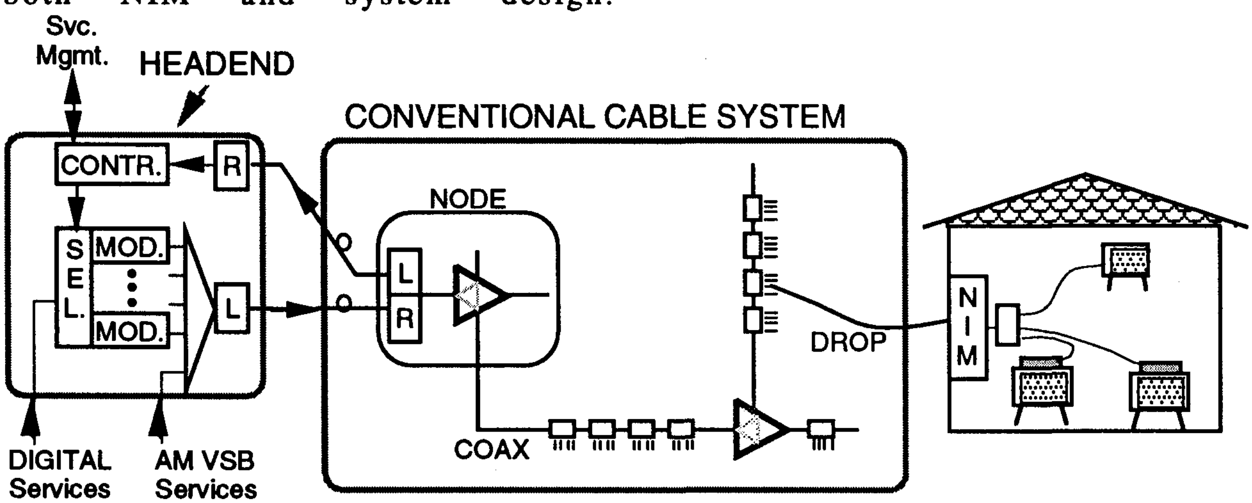


Figure 1. Cable System Overview

Integrated circuit 64 / 256-QAM or 8 / 16-VSB modulator outputs are combined and, together with AM VSB CATV Channels, passed to a laser for fiber transmission to its associated node. Upstream control signals received from the node provide, with negligible delay, control of selection of services into each downstream digital band. For requests involving authentication, usage billing or parental control, the control messages are passed via a local or centralized service management function (not shown).

The node would typically serve 600 homes passed. Line extenders must be fitted with 5-30 MHz (or 5-40 MHz) return amplifiers. Taps are preferably 1 GHz types to provide for future broadband upstream capabilities.

The home wiring should remain unmodified except for the insertion of the NIM between it and the drop. Domestic wiring and splitters should, in almost all cases, provide sufficient performance for the robust and level-insensitive DBPSK transmission.

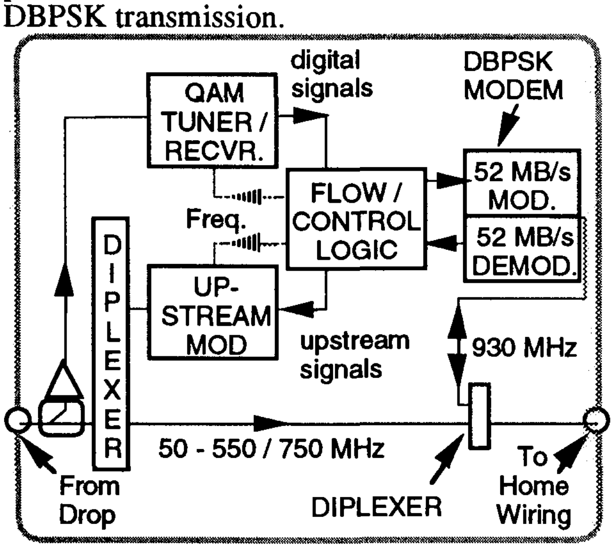


Figure 2. Network Interface Module (NIM)

The NIM illustrated in Figure 2 uses a diplexer to provide low-loss through routing of signals between 50 and 550 (or 750) MHz to the home wiring. Signals in the band 5 to 40 MHz from the upstream modulator are fed to the drop. An active directional coupler is used to tap the signal from the drop and feed this to the 64 / 256 QAM tuner/receiver This receiver contains low-phase-noise tuner, microreflection echo canceler, demodulator and forward error correction (FEC) functions. In traditional configurations these costly functions are needed in every set-top box rather than per home or MDU.

The flow control logic provides simple buffering between the QAM receiver bit-rate (27 Mb/s, 38 Mb/s or higher) and the 52 Mb/s TCM burst rate used in the in-home wiring. This function also inserts polling signals to trigger responses from set tops, etc.

The simple DBPSK modem provides upstream and downstream home wiring transmission. The associated passive diplexer is designed for very low loss below 750 MHz to avoid loss in the AM VSB path to the TV sets and provides the 930 MHz bandpass filtering for the DBPSK modem. The diplexer offers high through-path attenuation below 800 MHz to minimize unwanted upstream flow of in-home DBPSK signals into the QAM receiver and/or the drop.

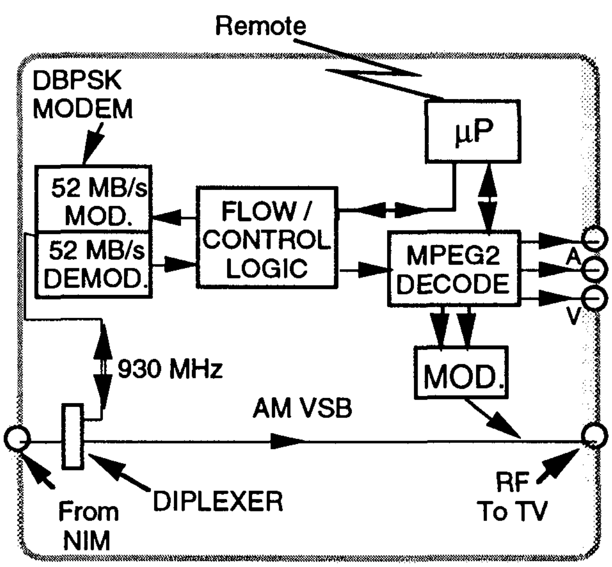


Figure 3. Low-complexity Digital Set-Top Box

The digital set-top box shown in Figure 3 uses diplexer and DBPSK modem functions similar to those used in the NIM. The flow logic buffers, selects and formats signals for the MPEG2 decoder and messages for the microprocessor. The flow logic also buffers upstream messages and responds to polling from the NIM. Other functions shown are similar to those in used in conventional set-tops. The most significant cost improvement over conventional designs is due to the absence of tuner, echo canceler, complex demodulator and FEC functions.

Options exist as to how to configure the set top box. Assuming that most "cable ready" TV sets tune all of the analog channels on cable, it is possible to add the output of the modulator in figure 3, to the incoming cable spectrum. This would likely be done either at the high end of the spectrum or at channel 14 (120-126 MHz). In order to overcome the noise on the incoming cable, it is desirable to supply the output of the modulator above the expected levels from the cable plant. This means that one channel must be unused between the modulator output and any analog channels, to prevent adjacent channel interference and to permit compliance with FCC rules. (The modulator channel is also not used in the downstream plant. This configuration is optimum, as it permits any watch and record scenarios, including those involving the digital signal. Alternatively, a switch to select either the modulator or incoming AM VSB band, is possible. An isolated MPEG2 RF output (in addition to the switched output) is also a possibility.

If it is to be assumed that a significant number of TV sets will not tune all of the cable channels, then a configuration which includes a basic or descrambling set top function is desirable. This raises costs and limits the ability to watch and record. However, since the decoded output is available at baseband, it is possible to route the descrambled signal to a VCR while watching something else, or vice versa if baseband inputs on the TV are available. With basic converter functions built in, it is possible to include a watch and record switch at low cost.

## Service Capacity

A significant advantage of assignment of QAM digital bands to digital subscribers, rather than to services, is that a very wide range of services can be provided initially and while subscriber digital service-take builds. This arrangement offers pay-as-you-grow headend and fiber/coax plant upgrade investment.

The curves shown in Figure 4 indicate average capacity per home served. Curve A applies to one laser feeding nodes offering service to 2400 homes passed. Curve B applies to a laser per 600 home-passed node. Curve C offers extremely high digital capacity which may be achieved using a separate fiber per node output port, i.e. serving 150 to 250 homes passed.

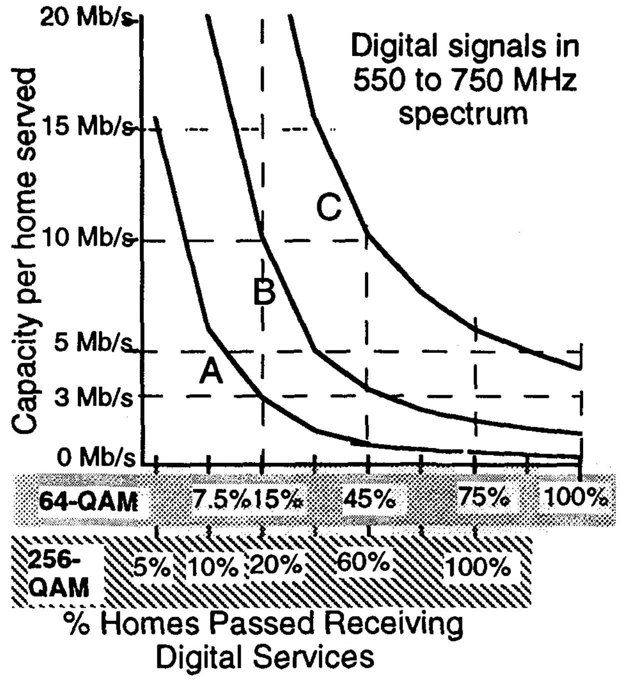


Figure 4. Digital Service Capacity

The two horizontal scales indicate the relative capacity of commonly proposed digital modulation schemes. Using silicon technology a single QAM modulator and up-converter will cost less than a typical QAM tuner/demodulator. Between 6 to 12 homes could be served from each modulator, according to service and video quality requirements.

## LAB AND FIELD TRANSMISSION VERIFICATION

### Lab Testing of digital demodulator robustness

Initial lab testing of the comparative robustness of a simple, non-coherent DBPSK demodulator and relatively costly, tunable, 64-QAM receiver with echo canceling and FEC indicates close to expected performance for each type.

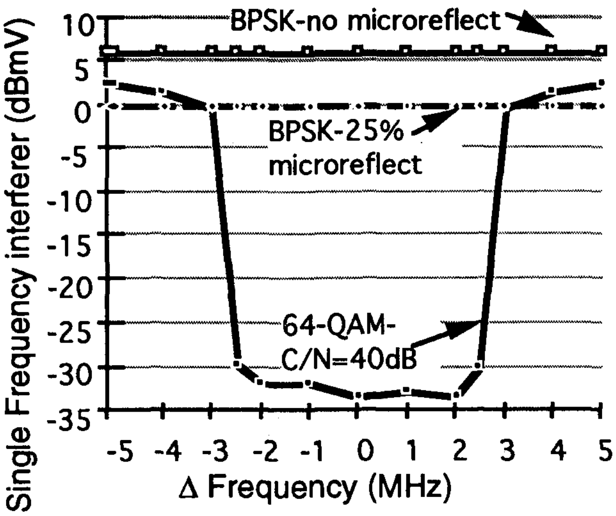


Figure 5. Comparison of QAM and DBPSK receiver single frequency interferer immunity

Figure 5 shows, on a ±5 MHz scale centered on each receive frequency, the comparative immunity of a simple, non-coherent 52 Mb/s DBPSK demodulator (without microreflection echo canceler (EC) or FEC functions) with 27 Mb/s 64-QAM receiver (with EC and FEC) for single frequency interfering signals in a 40 dB CNR environment. These tests indicate that the simple DBPSK approach is well over 30 dB more tolerant of single frequency interferers than that of directly received 64-QAM.

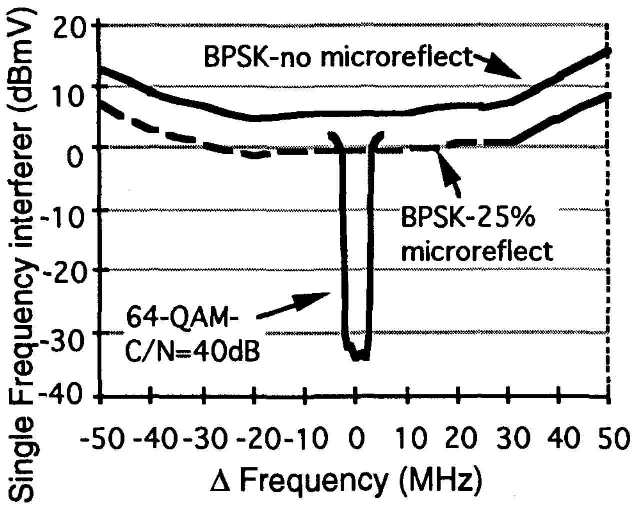


Figure 6. Comparison of QAM and BPSK receiver single frequency interferer immunity

Figure 6 shows the same data plotted on a ±50 MHz frequency scale and confirms the comparative robustness of DBPSK transmission over its full bandwidth. Note that the carrier level of the TCM DBPSK signal is higher than that of a typical, directly received, 64-QAM signal. This is acceptable since its energy is spread across a wider bandwidth.

Tests of the compatibility of the higher level DBPSK signals with a variety of "cable ready" AM VSB TV sets indicate no visible artifacts even when pulsed DBPSK signal levels of 30 mV are applied to the input of "cable ready" TV sets receiving 1 mV AM VSB signals. The immunity of analog set tops to this signal is now being verified. Set tops are expected to be even more tolerant of DBPSK signals due to their generally tighter specifications.

Field Transmission Plans

The comparative robustness of the two-way TCM DBPSK in-home wiring signals relative to that of directly received 64-QAM and 256-QAM is, at the time of preparing this paper, about to be tested in a selection of customer homes.

Signal generators will be used to represent interfering sources. In all cases the in-home wiring is not to be disturbed or improved. Subsequent tests are expected to confirm that known upstream home wiring ingress and egress issues have been fully resolved.

The spectrum of signals in each home will be monitored to identify degree of transmission margin remaining when interfering "cable-ready" TV sets are tuned across a range of AM VSB channels. Bit error rate testing will be performed in the presence of electrical interference and use of cellular handsets. Tests will also include consumer observations of the insensitivity of TV sets and analog set tops to much higher level than normal DBPSK signals.

Overall headend to Digital Set-top BER tests will precede tests using pre-encoded, stored, MPEG2 video. The video tests, although somewhat subjective, are expected to endorse the relative robustness of TCM DBPSK transmission in home wiring.

## HIGHLIGHTS

The known upstream and downstream transmission and system issues which are likely to impede introduction of digital services in the home can be resolved by the use of simple, robust TCM DBPSK transmission in home wiring.

Gating and buffering of upstream signals in a Network Interface Module before they enter the coax feeder network ensures that interference or faults in any home cannot affect the service of others. This also simplifies fault location procedures for the cable company.

Selection of digital services at the headend, rather than broadcast, offers economic service growth and an unlimited program variety including NVOD and VOD.

Use of headend selection also offers an attractive common service platform for a variety of non-TV services, particularly that of broadband multimedia work-at-home.

We believe that the proposed digital cable architecture has the potential to help MSOs become Broadband Telecommunication carriers of choice for both residential and small business markets.

Laboratory testing has been used to verify the proposed transmission and system approach. Initial field testing in feeder and home wiring is about to commence.

The proposed and soon to be verified architecture should help keep the cable company in control, provide for maintenance and enable existing and changing home coax distribution to be used consistently for digital services.

A major benefit of the proposed digital cable architecture is overall cost reduction in a real-world home wiring cable environment, both when installing service and through lowering of operating costs due to fewer subsequent truck rolls.

# Challenges with Transmission of data over CATV networks

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*ABSTRACT*

*In this paper the architectural and design considerations for the development of a network capable of reliable data delivery over shared CATV media are discussed. Potential challenges encountered during the transmission of data over CATV environment are examined. The media access method requirements and some implementation challenges are looked at and analyzed. The coexistence of data, telephone and video services and deployment considerations are also presented.*

## INTRODUCTION

Computers are becoming an indispensable item in the US household and business environments. Along with the presence of computers comes the need for their inter-connection to other computers, data banks and information servers. The internet has become the world's largest network, allowing people from almost anywhere in the world to communicate with each other and exchange information. Along with this capability comes the need for inter-connection at higher speeds. On-line service providers and users have always been limited and frustrated by the bandwidth of the telephone modem. Application programmers have also suffered from this, with their creativity being limited by the bottleneck.

The cable plant is inherently capable of providing very high communication speeds in a rather asymmetric fashion. There is much more bandwidth available in the downstream (to the home) direction than there is in the upstream direction. This asymmetry is surprisingly suitable for the home and small business environments, since the users predominantly retrieve much larger amounts of information than when they send. This property of the cable plant creates a very interesting opportunity for the cable owner, however, there are many challenges associated with providing this capability that we will try to address in this paper.

The first task is to choose a topology that will lead to easy operations and most efficient use of investment. Figure 1.0 represents a likely network topology for the delivery of data services. The network consists of three distinct physical blocks. They include the Master Head-end (MHE) equipment, Distribution Hub (Head-end) equipment and the Home unit. The topology of this network enables the operator to centralize the operations and the location of information serving computers, thus lowering operational and capital costs.

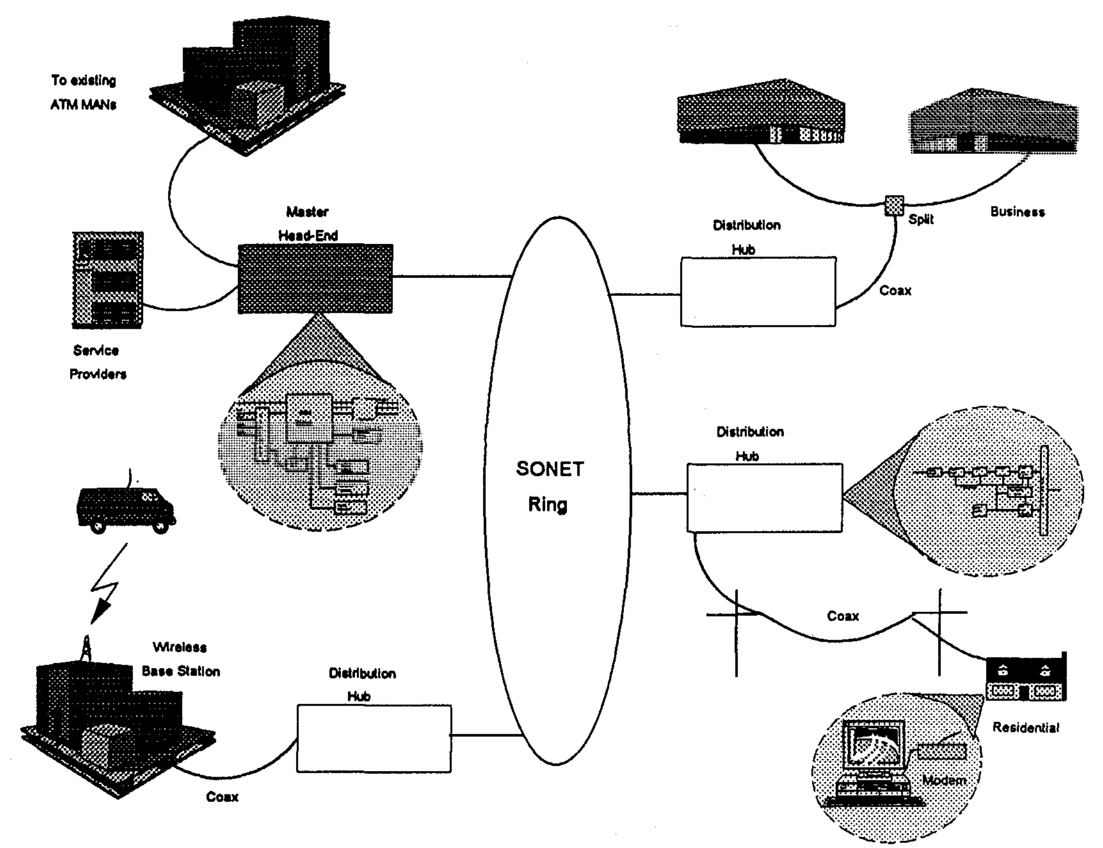


*Figure 1. Network Topology*

Among the functions likely to be performed at the MHE are billing, level of service authorization, and network operations control. The MHE supports several Distribution Hubs and therefore is a logical point to perform inter-hub switching or routing, as well as serving as a gateway to external networks.

The intended use of this network involves enhancing existing applications such as work-at-home, on-line services and access to the Internet and many other services that will be created as a result of the availability of large bandwidth. The home device should compete effectively against existing modem technology by allowing a user to avoid the cost of a second phone line dedicated to a modem connection and by enhancing the connection speeds from the typical 14.4 Kb/s to a 1.5 Mb/s rate. The connection between the computer and the modem should preferably be a standard interface (such as 10BaseT Ethernet) to allow ease of connectivity.

The product should also create new opportunities for services such as low cost video-teleconferencing, shop-at-home, and interactive games. New services for on-line applications will include video-clips and access to libraries of CD-ROMs.

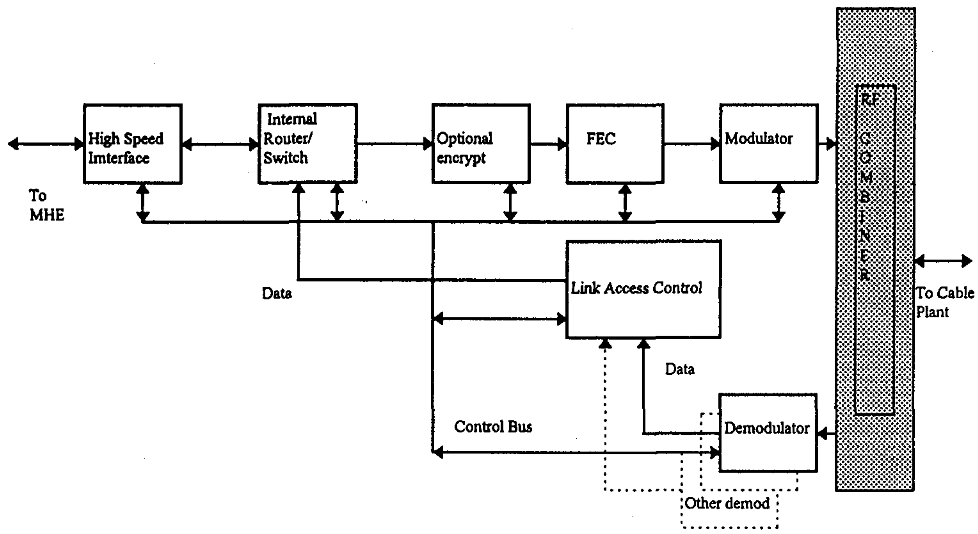


*Figure 2. Distribution Hub block diagram*

The Master Head-End may connect to several distribution hubs in a large metropolitan area. For smaller systems the Master Head-End and distribution hub may be combined at a single location.

A distribution hub (or head-end) houses a modulator that sends information towards the home using one or more RF modulators (Figure 2). There may also be several upstream RF demodulators to handle the return data. Since many sources may attempt to transmit upstream simultaneously on the shared return path, a Link Access Control mechanism must perform the control of the access method.

The customer premise unit resides at a home or small business and connects on one end to the cable and at the other end to a computer via a standard interface. The modem acts as a bridge between one or more computers and screens the traffic to decide which packets need to be forwarded onto the cable network. After deciding that the destination does not reside locally, the bridge terminates the Medium Access Control (MAC) frame and generates cells and forwards the information to its destination.

## USER REQUIREMENTS

The following is a list of requirements expressed by various network planners:

1. Provide LAN-like performance and connectivity as if you were at work.
2. Compatible with a variety of computer platforms and operating systems
3. Ability to connect into the wide area environment
4. Support TCP/IP and connectionless services.
5. Ability to charge based on differing service levels provided.

6. Eventual support of video-conferencing.
7. The upstream and downstream links need to be secure to prevent unauthorized use. Design of an end-to-end encryption scheme is an application layer problem and only the last link needs to be protected.
8. Support voice for use with interactive games.
9. Low cost for residential applications
10. Provide frequency agility to be able to work with other upstream devices and circumvent noise.
11. Prevent a single user from monopolizing a link.
12. Avoid creating another dedicated network management computer. Run as application on existing platforms.

For the optimum solution over the cable plant each of the requirements above must be closely examined. Most of the above requirements are realizable over existing baseband high-speed networks. The challenge is introduced when the same performance level is expected of the cable plant. There are a number of characteristics unique to the cable plant:

1. The length of the plant spans many miles. Distances of 15 miles from the distribution hub to the neighborhood are quite typical.
2. The plant is highly asymmetric. The available downstream bandwidth capacity is at least ten times greater than that of the upstream
3. The upstream reverse path is highly noisy (ingress)
4. The upstream bandwidth is needed by many services, including telephony, data, interactive video and energy management.
5. The point to multipoint cable plant topology

## DATA TRAFFIC CHARACTERISTICS

In order to design a network capable of providing such services, a detailed study of the traffic characteristics is necessary; it is important that the service provider have a good handle on the traffic growth over a period of several years. The traffic growth presents significant cost and planning implications over the Distribution Hub and MHE and cable plant equipment.

There will probably be several distinct types of data traffic carried to and from the home. The mix of these types will be variable by user, time of day and system. The product should support:

1. On-line services
2. Games
   - Interactive between several users
   - Game show response by a large number of users
3. Video-conferencing
4. LAN emulation for work-at-home applications

Each of these traffic types imposes different requirements on the transport network to ensure the customer receives the desired quality of service. These are given in Table 1.

| Type | Peak Down-stream | Peak Up-stream | Characteristic | Access | Latency | Jitter |
|---|---|---|---|---|---|---|
| Video conferencing | 384 Kb/s | 384 Kb/s | constant | fixed | fixed | low |
| Games (game show) | 50 b/s | - | extremely bursty | large | large | large |
| Games (interactive) | 1 Kb/s | 1 Kb/s | bursty | 15 ms. | 15 ms. | large |
| LAN Emulation | 1.5 Mb/s | 1.5 Mb/s | bursty | 100 us. | 3 ms. | large |
| On-line Service | 1 Mb/s | 2 Kb/s | bursty | medium | medium | medium |
| CD-ROMs | 720 KB/s | - | bursty | medium | medium | medium |

Table 1. Performance requirements for supported data streams.

## REQUIREMENTS FOR THE ACCESS METHOD

The upstream path is a very bandwidth-limited channel that must work to support the simultaneous needs of the upper layer protocol in transporting a variety of traffic types over a noisy link. The challenge is to provide a means for giving a user all they request at the quality needed, sharing the media amongst many users, and providing the statistics and diagnostics to ensure its reliability.

A connection can be described in terms of bandwidth (bits/sec) needed to be transmitted and received, latency (time to get access to the media), delay (time from sending to receiving), and jitter (variation of arrival times). As can be seen in Table 1, the various applications have different combinations of bandwidth, latency, delay and jitter and any combination of these may be present simultaneously. These new applications challenge the existing access methods to provide effective transport capabilities.

Traditionally, Local Area Networks (LANs) have not worried about these issues. They typically operate by having a station request access, grab the medium and use it for a packet time and then release the medium. There is no guarantee of the amount of bandwidth that a station can get and no way to keep a user from "hogging" the available bandwidth. Because of the way LANs operate, it is not possible to have a deterministic connection.

The most common access method for LANs is known as Carrier Sense Multiple Access with Collision Detection (CSMA/CD) which is a technique that listens before transmitting and listens while it is transmitting to ensure that no other station interferes. For a lightly loaded LAN (30%) there is a good probability that the LAN can handle time critical data. As the link utilization increases, the probability of a collision increases and re-transmissions become necessary. This means that the receiver will see variations in the scheduled arrival time (jitter). Long packets will also cause jitter because a station may want to transmit, but it must wait until the current packet has completed transmitting. The longer the allowable packet length the greater the potential jitter.

Providers of this service would like to offer a tiered service which would be priced based on resource utilization. A higher tiered connection might be a video-conference at 384 kb/s and a low tiered service might be an interactive game. This requirement of access method means that both a minimum guaranteed bandwidth and a maximum bandwidth limit must be placed on a stations connection(s).

The last major requirement of the access method is it that it must operate efficiently. The metrics of efficiency are a balance between maximum link utilization (bits/second transferred), cost ($/mega Hertz), and bit error rate (BER). Overhead for a Forward Error Correction (FEC) code takes away from the bandwidth but reduces the need for retransmission. More complex modulation schemes that are capable of providing more bits per Hertz are desirable but are more costly and have a higher BER for a given signal-to-noise ratio (SNR).

Work is being done in various standards committees to solve many of these issues. This work is aimed at enabling product offerings from many manufacturers, thus forcing down the equipment prices. Among the organizations looking at these issues are the IEEE 802.14 committee, ATM Forum and DaVIC.

Among the many possible solutions being considered are modifications to the IEEE 802 protocols so that an application can negotiate the characteristics of the needed connection with the network. Centralized control of station access will be required to ensure that a user does not utilize the shared link beyond its authorized service level. Security will be needed on the upstream and downstream links to ensure that other stations cannot use information destined for another station. Forward error correction must provide a link performance of 1 error in $10^8$ bits.

## NETWORK LOAD PROJECTIONS

The following is the projected analysis of the amount of data that the network will be expected to handle. The assumptions are:

- 100,000 subscribers /cable system
- 5 hubs / master head-end
- Erlang calculations are based on 1% call blocking probability
- 64-QAM @ 27 Mb/s
- Burst mode QPSK upstream @ 1.5 Mb/s
- Downstream link utilization 95%
- Upstream link utilization 80%

| Year | Per Distribution Hub Downstream / Upstream | | Per Master Head-End Downstream / Up-stream | |
|---|---|---|---|---|
| 1 | 6.5 Mb/s | .8 Mb/s | 33 Mb/s | 4 Mb/s |
| 2 | 11.6 Mb/s | 3.2 Mb/s | 58 Mb/s | 16 Mb/s |
| 3 | 23.7 Mb/s | 7.3 Mb/s | 119 Mb/s | 37 Mb/s |
| 4 | 76 Mb/s | 25.9 Mb/s | 380 Mb/s | 130 Mb/s |
| 5 | 116 Mb/s | 45.6 Mb/s | 580 Mb/s | 228 Mb/s |

*Table 2. Network Load Projection*

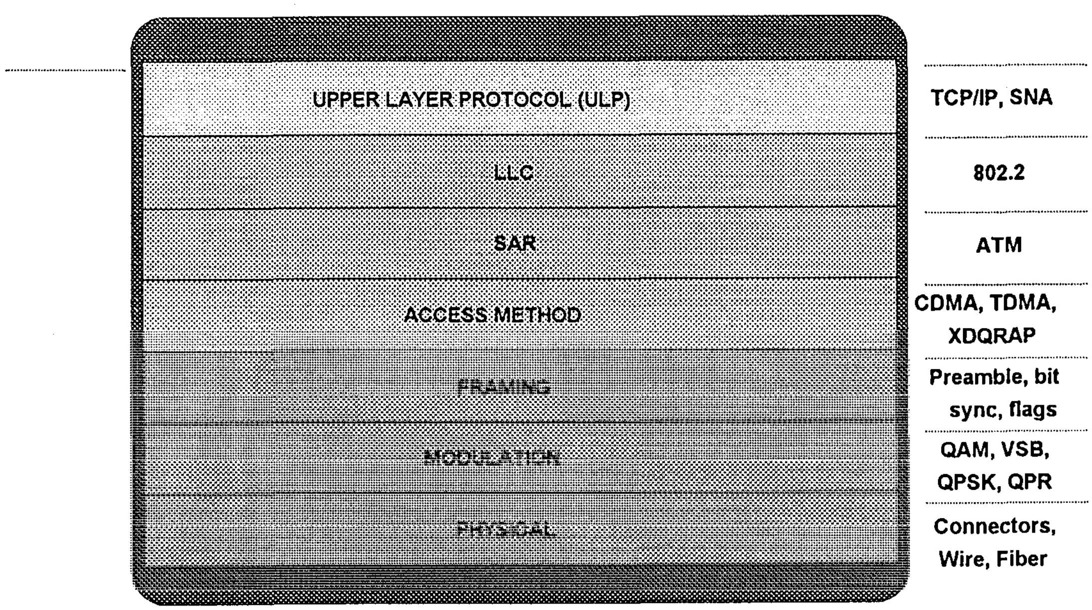


*Figure 3 Protocol Layers*

## REVERSE PATH ISSUES

### Working in a noisy environment

The existing cable plant was not originally optimized with the intention of one day providing the interactive video, data and telephony services. This has created a challenge for the equipment providers and service providers. The ingress from off-the-air signals into the drop is one of the largest sources of interference. Energy is coupled in through open terminations and impedance mismatches at the drop cable and is funneled through to the head-end receivers. Other sources of interference and impairments are thermal noise and non-linearity of the reverse active equipment.

Ingress corresponding to signals from short-wave broadcasts and two-way radio signals is observed as relatively stationary narrow-band interference on the cable return. Transient, impulsive impairments are caused by both natural and man-made sources such as lightning, industrial and home appliance noise sources.

The reverse link must be able to maintain a minimum level of performance ($10^{-8}$). Long term performance is indicated by Bit Error Ratio (BER), which is the probability of observing a bit-error over a very long observation time. In a data transmission world, large BER's result in data re-transmission or failure. The effect on the user is the perception of very slow response times.

A robust cable data transport solution must address both of these issues:

1. The stationary interference is overcome by building dynamic frequency agility into the reverse modulators and demodulators. The problem becomes more challenging as the reverse bit-rate increases, since the number of potential users and the amount of proportional occupied spectrum increases. At T1 rates (1.544 Mbps), approximately 1 MHz of spectrum has to be available to allow dynamic re-tuning. Building cost-effective hit-less frequency change-over schemes is challenging.
2. The impulse hits can be overcome by either using Forward Error Correction (FEC) schemes on the data or by just putting the burden onto higher layer protocols such as Transmission Control Protocol (TCP). Forward Error correction techniques can provide correction in small blocks of data. These blocks could be as small as one symbol. Thus, if the impulse hits are relatively long, FEC will not help and other alternatives must be used.

### Coexistence with other Services

Currently the only available scheme for enabling coexistence of various service types on the RF plant is for each service type to own its own unique upstream and downstream channels. By using this inherent property of the cable plant, we can guarantee coexistence. Consequently, it is necessary to know the RF bandwidth requirements of each of those (digital) services in each direction of traffic, and their required quality of service parameters, such as tolerance to various types of errors, delay and jitter. Also, knowledge of the RF actives and passives, specifically their gain versus frequency, their phase (group delay) versus frequency and gain versus level behavior is necessary, of course during initial system commissioning and later for dynamic transmission control. The ability to control the power levels of the various reverse transmitters for optimum signal reception and minimum interference is another challenge. A high level carrier from inside the home may have undesirable distortions on the lower end of the forward spectrum and potentially overdrive the reverse amplifiers and optical lasers. On the contrary, too low of level will not provide the needed Carrier-to-Interference ratio. Certainly, the future migration of various service elements at the home into a more integrated platform will alleviate some of these problems.

## NETWORK MANAGEMENT

According to the International Standards Organization (ISO), network management consists of five subsystems, Fault management, Configuration Management, Accounting, Performance Management, and Security.

Network management must be viewed as an enterprise requirement and not just a product requirement. The effective management of an enterprise network requires easy and open interfaces to each network element managing agent, intelligent knowledge of the capabilities and needs of each network element and finally the knowledge all network element inter-dependencies.

The common management protocol, being currently considered by some industry leaders, is the Simple Network Manager Protocol (SNMP). SNMP is part of the TCP/IP protocol suite. SNMP provides a common mechanism for the communication of management information among network elements and element and network managers. Some the required activities for the proper management of a network were alluded to in the previous section. The discussion of reverse path spectrum management and cable plant management fall under configuration management and performance management.

A network management platform consists of:

### Fault Management

Fault management is the process of isolating faults within a network and reporting the faults and the cause of the faults to the appropriate staff. The Network Management Platform (NMP) should prioritize all critical data and filter all non-critical data. This task requires the NMP to understand the network topology and correlate events from a number of different element managers.

### Configuration Management

Configuration management consists of all activities related to the management of critical resources. For the cable plant, these activities include the provisioning of all modulators, demodulators, RF spectrum and amplifiers as well as digital equipment.

#### *Spectrum Management*

Spectrum Management is a form of Configuration Management unique to the cable plant. As soon as two-way digital services are deployed on the plant and as soon as reverse traffic start to build up, so will the need for intelligent management of the spectrum. The effective management of the reverse spectrum to assure efficient utilization, management and cohabitation of services in the reverse link, will enable the service provider to maximize its revenues and minimize its operational costs. The ideal reverse spectrum management system is capable of monitoring the performance of the reverse spectrum and dynamically making optimum spectrum assignment decisions. The ideal "Reverse Spectrum Manager" must be able to retrieve link performance data in both directions from the various RF transmission equipment on a given plant, know the performance and RF bandwidth requirements of each service type, and be able to command the modulators and receiving demodulators to re-tune to various RF channels at the optimum power levels.

### Accounting

Accounting consists of the entire process of service request authorization, billing and bookkeeping. The accounting susbsystem must maintain a complete and up-to-date list of authorized users, against which service requests are compared, validated and authorized.

### Performance Management

Performance management is the process of analyzing the quality of the network. The data gathered allows the network manager to be proactive in alerting maintenance staff prior to any system degradation which could affect service.

### Security Management

Since network management serves as a platform with full control and access to critical system functions and data, system access security is a key factor in the design of any network manager. The NM must give the system administrator the ability to flexibly assign privileges to system users.

Another aspect of security management falls in the realm of signal security and access authentication. This aspect is of crucial importance in a cable topology. The downstream broadcast nature of cable, as well as ease of physical accessibility makes the plant vulnerable to unauthorized users. Scrambling and encryption are techniques being used today to provide conditional access and prevent eaves-dropping. The ideal network management system would have full control of all accesses and will be able to detect unauthorized accesses.

## CONCLUSION

In this paper we presented some important issues concerning the offering of data services over cable TV. Network scalability for ease of growth, robust reverse access and modulation schemes, spectrum management, secure links, interference management and in general network management are of utmost importance in the planning of these multi-media networks.

# Conditional Access and Security Considerations of Transactional Broadcast Digital Systems

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Time Warner Cable

*Abstract*

*The Orlando Full Service Network (FSN) will be one of the earliest deployments of a true digital interactive broadcast network. Most conditional access systems found in LAN/WAN applications assume an "honest" client. As we move deeper into consumer entertainment, this conventional model demonstrates its limitations. This paper explores the technical and operational considerations for the conditional access system's requirements in an highly interactive environment.*

## INTRODUCTION

A conditional access system is a security system that will allow a customer to use all services that they have purchased, and deny them access to services they have not purchased. Typical services that are protected in cable television are; premium channels, tiered channels, PPV, and more recently, digital music and Sega channel. Security is the protection of sensitive information that is communicated over the cable system. Types of items that fall under security are; password protection, code download, credit card verification, PIN numbers, e-mail etc. Both conditional access and security are now becoming issues that must be dealt with in cable systems as we began to deploy distributed, transactional networks.

### Conditional Access

Presently, conditional access is a function that is handled by converter manufacturers and billing system vendors. When a customer orders a service, a customer service representative (CSR) inputs this information into the billing system. The billing system logs this input into a database and sends an instruction to the converter's headend computer. The headend computer relays this instruction to the converter via an encrypted data stream in the vertical blanking interval of a video channel or in an "out of band" signaling channel. The encrypted instruction tells the converter what channels it can tune and descramble, and what functions it is authorized to perform. PPV is a special case of this authorization process. The difference between PPV and premium channels are that for PPV services most converter are pre-authorized with a defined number of event credits. When a subscriber orders an event, the converter authorizes itself for the event, if the subscriber has any credits. The converter then stores the knowledge that the event has been purchased until it is polled by the headend computer. These traditional types of services only involve the converter manufacturer, the billing vendor, and a CSR when a customer is changing levels of service such as adding or deleting a premium channel.

On an interactive network, conditional access has to be handled by a number of different vendors. If a customer orders a premium channel or a PPV event, this has to be handled through the analog converter's conditional access system. This can be handled the same way it is today. But what about other services such as interactive games and VOD? These services aren't delivered through the analog portion of the set-top, they will be delivered through the digital portion of the set-top. To order a movie, the subscriber needs to get access from the VOD provider. To order a game, one would need to get access from the game vendor. The game may be billed independently from the cable bill. This creates a situation where one may have to get a credit card verification in order to play a game. This transaction would require the subscriber to get access from the game provider and the game vendor to get credit card verification. Both of these actions require that a secure path be provided from the digital portion of the set-top to the service provider or to the billing system. But neither of the actions listed involves the analog portion of the set-top. Conditional access to these services will have to be controlled by an entirely different method.

As demonstrated above, the billing vendor and converter manufacturer are not the only vendors who are involved in the conditional access function on an interactive network. There will be many other service providers that need to give or deny access to their products. This array of services and different vendors forces one to take a different approach to control access in an interactive network. It also points out that current analog conditional access systems cannot be used to control access to digital services since they aren't involved in the process.

The interactive networks that are envisioned are more like the LAN/WAN environments that exist today. These environments are distributed computing environments. On these networks there are many clients, or customers. There are also many servers, or application providers. These networks have implemented their own type of security. One popular method that is used to secure a LAN or WAN is to use a Kerberos authentication system. This system uses DES encryption in conjunction with user logins and passwords to protect the network.

Unfortunately, it is difficult to force our subscribers to use logins and passwords to gain access to our system. To make this type of a security system viable, the access control needs to be transparent to the users. This forces one to put the login features inside the set-top in a cable environment. But, most LANs and WANs don't use much "physical security". The security algorithms that are in use are software programs that run in memory on the machines. The real security in this type of network is in the secrecy of the user's password. It is assumed that users won't give their passwords to others since they will be held responsible for the other person's actions. This type of protection isn't applicable for a cable system. There will be many services that are available that could be used without authorization in this type of environment. But, without some type of physical security the set-tops that are deployed in a cable system could be cloned and sold to dishonest customers so that they could gain free access to cable services.

The security system that is needed for interactive networks needs to take the features from both of these networks and combine them. The physical security of traditional set-tops is needed to simplify the user interface, and the multiple services to multiple customers paradigm of the distributed computing environment is needed to allow for a rich set of new services.

## INTERACTIVE NETWORK SECURITY

On an interactive network there needs to be a secure way for messages and commands to be sent across the network to and from subscribers, operating support systems and service providers. The security system should be transparent to every one operating on the network. To build this type of security system, security measures must be embedded into the interactive network itself. This implies that all of the transactions on the network be secured.

There are some fundamental tasks that a security system must perform on an interactive network. They are:

- Protect privacy of messages
- Provide for conditional access
- Protect against viruses

- Protect against illegitimate kernels and application binaries
- Protect against cloning

FSN Security

Security was embedded into the operating system of FSN. When the FSN set-top is turned on, the set-top only has a small bootstrap loader resident in read only memory. All of the software that runs on the set-top, except the bootstrap loader, is downloaded to the box after it is powered up. This gives one the flexibility to change the operating environment and the user interfaces for the set-top at any given time. Due to the fact that the box is totally downloaded with software, it was necessary to begin implementing security procedures the moment the box is turned on. This is the first step in security in the FSN.

The first task that the bootstrap loader does is to get the boot parameters file for the set-top. The boot parameter files is a data file which is being constantly transmitted over one of the digital channels in the neighborhood. The boot parameter file contains all the information the set-top needs to know regarding how to work on the network. The contents of the boot parameters files are encrypted with one of the bootstrap keys that is stored in the set-top. It is essential that the bootstrap key in the set-top be protected. If the bootstrap key is ever compromised, the set-top can be defeated since this key decrypts all of the other keys that are used in operation. There are several safeguards that are used to protect the bootstrap keys. First, the bootstrap key must be stored in a secured microprocessor inside the set-top. This is to guard against visibility to the keys. Second, to extend the life of the bootstrap keys, they are only used during power up to decode the boot parameters file. Third, the boot parameter files are only broadcast in the neighborhood where the set-top is authorized to be. If the boot parameters keys are ever comprised and the set-top is cloned, the cloned set-tops would only operate in the neighborhood where the original set-top was activated. Fourth, there are several bootstrap keys stored in the set-top's secure microprocessor. If one of the keys is compromised, then the set-top can be instructed to use a different key by changing the bootstrap key reference variable in the boot parameters file.

Once the set-top receives the boot parameters file it performs an integrity check on the file. The integrity check is another important security task. The purpose of the integrity check is to verify that the file that the set-top received was originated from a trusted source. The integrity check performs a secure checksum calculation over the boot parameters file. If the value of the secure checksum calculation matches the value that is stored in the boot parameters file, then it assumes that the boot parameters file is from a trusted source. After this important security step, the set-top has enough information to be able to download the operating software or operating kernel. The contents of the boot parameters file are :

Set-top ID,
Bootstrap key #,
$\%_{\text{(bootstrap key \#)}}$Boot channel frequency,
$\%_{\text{(bootstrap key \#)}}$Boot kernel name,
$\%_{\text{(bootstrap key \#)}}$Kernel integrity checksum,
$\%_{\text{(bootstrap key \#)}}$Bootstrap integrity checksum,
$\%_{\text{(bootstrap key \#)}}$Runtime key,
$\%_{\text{(bootstrap key \#)}}$Other needed data.

Note: %'s implies encrypted with key (bootstrap key #).

The set-top now knows the frequency to tune to download the operating kernel and the name of the file that it should download. The set-top now attempts to download the operating kernel. Once the operating kernel is downloaded, an integrity check is also performed on it. This is to insure that the operating kernel is legitimate, and not being sent by a pirate. The operating kernel and all application software downloaded by the set-top is sent through an integrity check. The integrity check performs a secure checksum calculation on the downloaded operating kernel just as it did on the boot parameters file. If the calculated value matches the value that is in the boot parameters file, the operating kernel is believed to be legitimate. It is then stored into memory and control is passed to the operating kernel.

After the set-top operating kernel begins to execute, it is in charge of security. Since all applications reside on top of the operating kernel, when a request is made to send a message, the message is automatically secured as part of the operating kernel's secure messaging function. The set-top's operating kernel always talks to the network operating system using secured messages encrypted with the "runtime" key. This is the first level of conditional access in an interactive network. Whenever an application requests network services, the operating kernel must make this request on behalf of the application. This process is called session establishment. To establish a session, the application passes a request to the operating kernel specifying the requirements of the session being established. The requirements includes information such as, who the session is with, the type of bandwidth required in the forward and the reverse directions, etc.. The operating kernel takes this request and then encrypts it with the runtime key that it received in the boot parameters file. The operating kernel then transmits the request to the session manger. Session manger receives this command and passes it to the authentication server. The authentication server is the only element in the network that knows the bootstrap keys for set-tops. The authentication server decrypts the message and passes the message back to session manager with a session key. The session key is the encryption key that will be used between the application server software and the set-tops application software. Session manager takes the information that it received from the authentication server, determines where there is enough bandwidth to support the subscriber's request, and creates a "ticket", (Figure 1). The ticket consists of:

Set-top ID,
$\%_{(\text{runtime key})}$session key,
$\%_{(\text{runtime key})}$forward channel frequency,
$\%_{(\text{runtime key})}$forward bandwidth
$\%_{(\text{runtime key})}$reverse application frequency
$\%_{(\text{runtime key})}$reverse bandwidth
$\%_{(\text{runtime key})}$ticket lifetime,
$\%_{(\text{runtime key})}$other relevant data.

Notice that the ticket has a limited lifetime. If a movie is ordered, the tickets life may be valid for 1.5 times the movie length. This ensures that the ticket cannot be copied and replayed to fool a set-top box. Session manager now takes the ticket, encrypts it with the set-top's runtime key, and transmits it back to the set-top. Since the ticket is encrypted with the set-top's runtime key, only this particular set-top can decode the message. Anyone who is listening in on the RF spectrum would not be able to tell where a particular set-top was told to tune. The ticket is also transmitted to the server using a secure key. This insures that service providers have to go through session manager, and billing, to get to a customer. Until session manger gives the server side of an application a ticket to a set-top, there is no way for a vendor to talk to a set-top. The application vendor must use the ticket's information and encrypt it with the session key before it can be recognized by the set-top.

This is how we perform network conditional access in the FSN, it is a part of the secure messaging function of the security system. Once the set-top and the server receive their tickets, they have enough information that

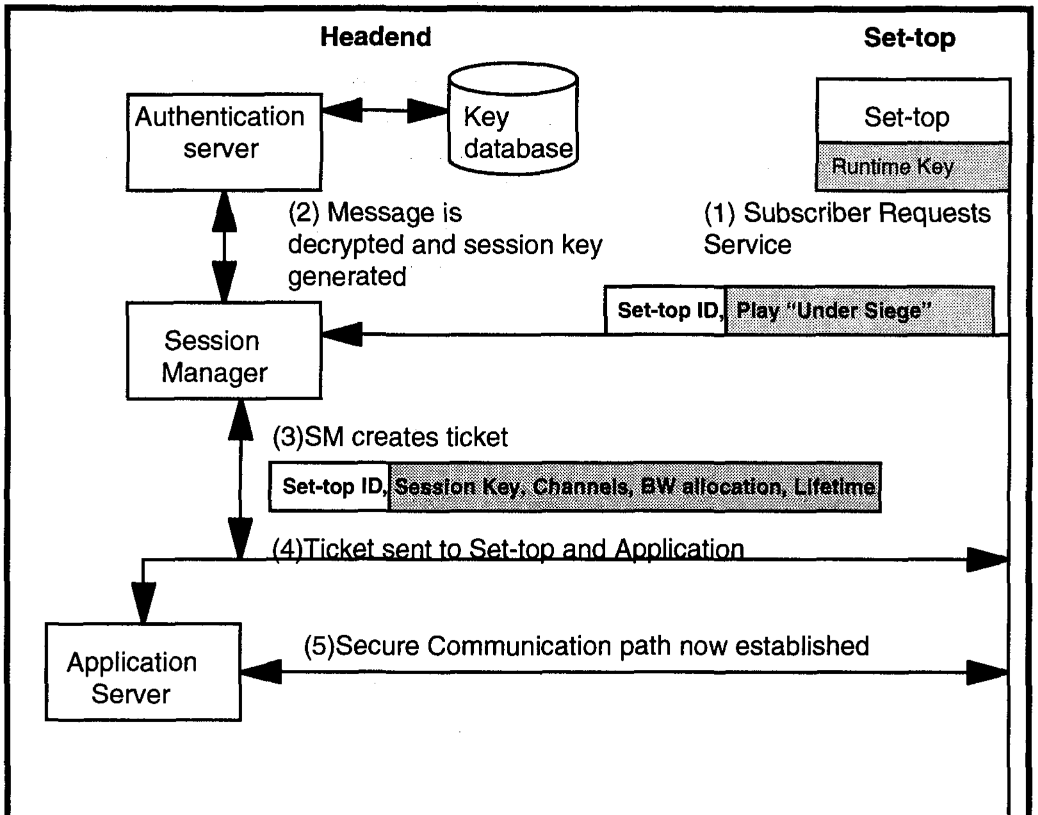


**Figure 1. Session Establishment in FSN.**

they can communicate with each other. Since the set-top will attempt to decrypt any message from the application provider using the session key, the application provider must use this method if it expects a set-top to understand the message.

Conclusions

The bottom line of any security system is how well it performs over time. This system has yet to be tested on the large scale like converter conditional access systems, but there has been a lot of thought given to how one might break the system. The FSN security does provide for many of the features that are required of security system, but there are some features that are missing.

First, the security provides for secure messaging. All messages are secured when they are transmitted over the network. This protects customers and vendors against snooping. Anyone snooping on the network will have to know the proper key in order to decrypt a message. It gives message privacy, since any application that attempts to talk to a set-top must be given a secure ticket in order to get network services. If an application needs to get credit card information from a customer, then they can be sure that only the application and the customer can decode the message. This also provides for the first level of conditional access since a ticket must be given by session manager before anyone can communicate on the network.

Second, it protects against viruses and bogus codes. Since knowledgeable pirates might try to fool a set-top with a phony applications that could potentially steal credit card information or other private information, the integrity check verifies that all software has come from a trusted source. If someone attempts to load a virus onto the network, this feature would also prevent it from being loaded and executed.

Third, the security system is seamless to applications and to customers. No one has to think about whether the transaction is at risk. All messages will be encrypted automatically with a session key. Since the session key is random and has a limited lifetime, it is difficult for a pirate to receive information and decrypt the information using powerful computers. By the time the key is discovered, it will probably be invalid.

And finally, the bootstrap key is protected in a secure microprocessor. The bootstrap key also has a limited neighborhood scope. It is only valid in the neighborhood that the set-top is authorized in. This should minimize attempts to clone a set-top since a cloned set-top would only be useful to the subscribers in a 500 home node. The life of the bootstrap key should also be extended since it is only used when the set-top boots. There are also multiple bootstrap keys in the set-top. If one of the keys is compromised another key can be used by simply changing the boot parameters files.

One of the problems with the FSN security is the visibility of the runtime key and the session keys on the computers bus. Even though these keys have limited lifetime to really make this a secure system these keys need to be hidden within the secure microprocessor. The only output of the secure microprocessor should be decrypted messages. Another problem is that currently MPEG video is sent in the clear. There also needs to be a high speed decryption path through the secure microprocessor for the MPEG video data. Since the security process described is software, there isn't enough processor speed to decrypt the video stream and to perform other needed functions. But this task could easily be done in secure hardware, since the video streams are typically running at less that 6 Mbits/Sec.

Will this be enough security to protect the increasing amounts of information that flows across future networks? Only time will tell, but if anyone has begun thinking about potential problems with the FSN security system, and ways to fix the problems then I have succeeded in my task.

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# CONSIDERATIONS FOR DEVELOPMENT OF EXISTING CATV NETWORKS FOR FUTURE TELECOMMUNICATIONS SERVICES

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A major portion of the current CATV systems in the United States have yet to be upgraded. A lot of work has been done on reducing amplifier cascades with fiber, and a few companies have started implementing their best guess of what a future Hybrid Fiber Coax (HFC) network should look like. Not to belittle anyone's efforts, but everyone is guessing! Almost every new service being proposed today has been talked about since the late sixties. The major drawbacks to implementing most of these services are both technical and sociological in nature. The Warner "Qube" project and at least one videotex experiment that I am aware of are examples of technologies deployed before their time. Over the last few years there have been several breakthroughs in fiber optics and communications technology, and integration in the computer industry has made many of these services technically viable. Much of the uncertainty surrounding how to build a HFC network centers around which of these services will be marketable and how much demand there will be for them.

From a technical point of view, the best answer to this uncertainty is to build a network which is flexible enough to handle any foreseeable situation. While this is very attractive to the engineer, it may not be very practical -hence, the educated guess.

All else aside, the cable industry has the unique ability to do two things very well: broadcasting and narrowcasting. Our competitors may be able to do one or the other, but not both with any reasonable degree of efficiency. Broadcasting multiple analog telecommunications signals has been our bread and butter. It will continue to be a very important part of our success in the future - after all, there are several hundred million analog TV sets in the US alone. With the advent of fiber and other related technical advances, we should be able to overlay an efficient and reliable high speed digital network in place, which will allow us to narrowcast individual services to our customers. Achieving this mix in an efficient integrated network is what will allow us to survive in the future. The remainder of this article will discuss several of the technical issues surrounding the requirements and concerns of providing these proposed future services.

## Architectures

A network must be both flexible and reliable enough to allow for future services. Most of the current HFC designs have a common thread. A service area is defined, based on a specific category such as homes passed, and a fiber optic node is placed in a convenient location to serve that area. The services are then distributed throughout the area using a coaxial bus. The size of the service area, the number of fibers allocated, the amplifier cascade allowed in the bus, redundancy, path diversity, noise and distortions, etc., are all factors that vary widely across the industry. No matter which version of HFC architecture that one is deploying, it is significantly better for distributing the traditional analog broadcast channels than a long cascade, as in the branch and tree systems of the past. Narrowcasting has become the driver for network design.

The concept of narrowcasting changes dramatically the way one must think about system design. In a traditional cable television headend, 99+ percent of the signal sources go to every subscriber. To narrowcast, one must be able to supply a set of unique services to each individual subscriber. Hence statistical traffic capacity analysis must be incorporated into network design. Also, if one believe that a lifeline telephone service is in one's future, and that it will be network powered, variable current and voltage loading concerns must be addressed. A separate section discussing powering issues will be presented later.

What we don't know about the services littering the information super highway is scary. What we must do is proceed on one's best educated guess. For example, Video On Demand requires a 3 to 5 Mb/s downstream capacity for each unique user, but only a bursty, very low data rate capacity upstream. Video telephony, on the other hand, will probably require a symmetrical 384 Kb/s per active caller. These are two of the most talked about services, and are also the biggest wildcards with respect to network loading. If one chooses a 25% peak loading in a 60% penetrated 500 home service area, VOD service will require approximately twelve individual 6 MHz wide Quadrature Amplitude Modulated (QAM) downstream channels: (4Mb/s *.25*.6*500 homes)÷27Mb/s. OK, but what if your penetration varies between 10% and 95%, or a particular neighborhood has an atypical number of VOD customers on a given day? There are three main choices: 1) Ignore the problem and eventually sell your system, 2) build a Rolls Royce network that will not run out of capacity but will cost so much you may never break even or, 3) build a system with the flexibility to change with the business as necessary.

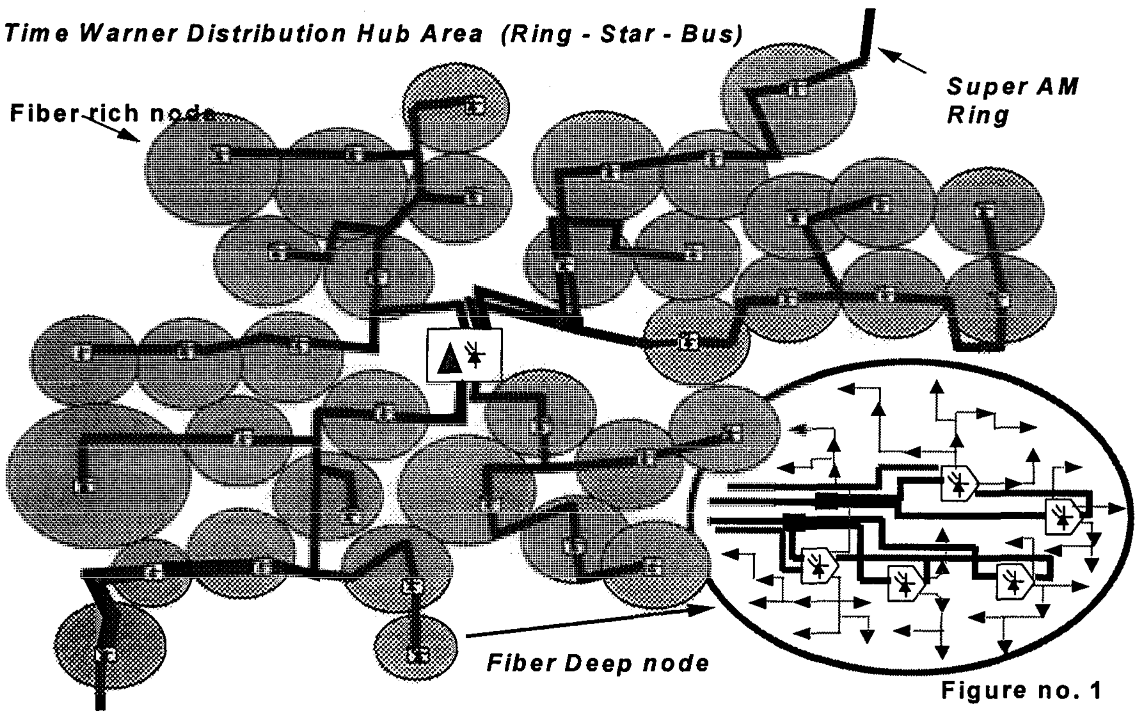

Figure no. 1

One of the first decisions to make is the size of the service area. Today, one's choice should be based on an acceptable set of analog video distortions, a reliability factor one can live with, and a cost effective migration path to a smaller service area that can be overlaid onto the existing network with a minimum of rework and customer disruptions. Our experience with reliability has been that as the amplifier cascade increases, reliability decreases on a logarithmic basis. On the other hand, a purely passive coaxial bus is very expensive and does not yield significant gains in reliability over a HFC design with short cascades. A recent paper on reliability(1) suggests that system availability differences between passive and near passive HFC designs are less than three minutes per year. The current difference in upgrade cost between these two approaches is several thousand dollars per mile.

Next, fibers are home run from service areas to the hub; enough fibers are used to provide flexibility. In many cases the labor expense of installing a fiber optic cable may cost more per foot than the cable itself. It also may pay to consider increasing the number of hub locations in the system. By shortening optical paths to the service areas, one can save "up front" on the cost of laser transmitters and high fiber count cables. In many cases, this monetary difference is more than enough to pay for the land, building, and electronics. It is also presents an opportunity to provide path diversity and electronic redundancy at a location which is reasonably close to the customer, without an exorbitant price penalty.

## FIBER RICH UPGRADE CPM ANALYSIS

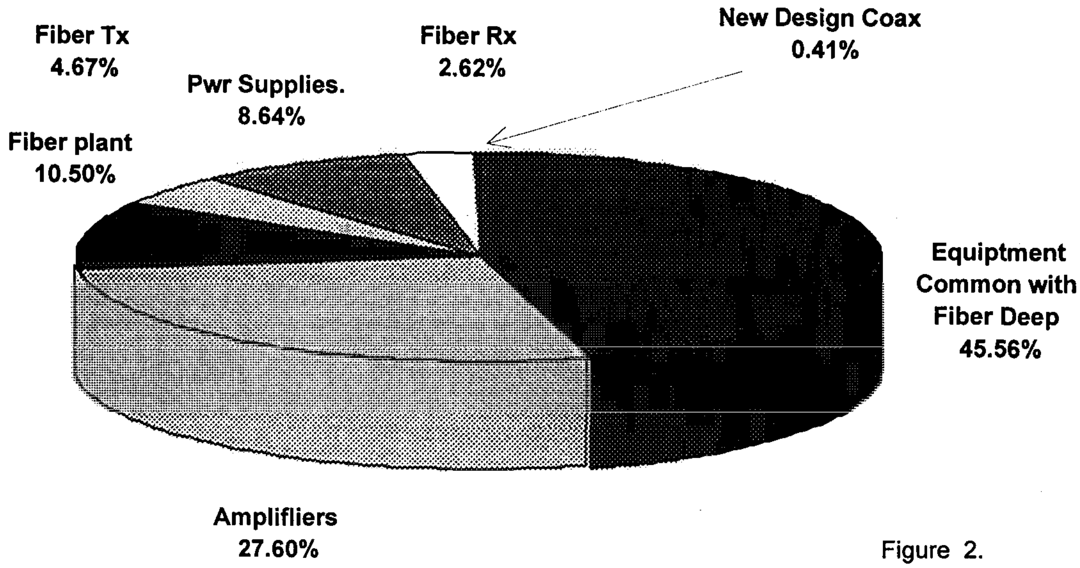


Figure 2.

Time Warner has developed a Ring-Ring-Star-Bus architecture as a compromise between current and future needs. The service area averages 500 passings and hubs serve roughly 40 service areas. This gives us very short optical path budgets and limits our exposure to loss of service from damage to the star(Figure 1.). Presently, Time Warner is using a coaxial bus architecture called Fiber Rich. This design calls for a maximum cascade of two trunk amplifiers, one bridger, and three line extenders. This cascade allows us to easily serve 500 passings where densities are greater than 50 hpm. This design has also been fairly cost effective, upgrading the typical 300 MHz system to 750 MHz using one GHz taps and passives with an average costs per mile in the 12 to 14 thousand dollar range (Figure 2.)

## Fiber Deep

Because the price of fiber optic cable and electronics has been steadily decreasing, we are now able to move fiber optics deeper into the existing coaxial plant. This new architecture is called Fiber Deep. To prove the feasibility of this new design, we performed cost analysis on several test designs, with various ratios of fiber to coaxial plant.

Fiber Deep is a modified fiber to the bridger architecture. Our analysis indicated that pushing fiber beyond this point in the plant was costly and provided very little improvement in efficiency or reliability. Fiber Deep, on the other hand, trades the cost advantages of fewer amplifiers and power supplies against increases in fiber optic cable and electronics.

| MONROEVILLE SERVICE AREA COMPARISON | | | | |
|---|---|---|---|---|
| | | | | |
| Service Area | Feet of coax | Miles of coax | Amplifiers | Amplifiers per mile* |
| | | | | |
| Fiber Deep #002 | 29389 | 5.57 | 30 | 5.39 |
| Fiber Deep #003 | 22965 | 4.35 | 24 | 5.52 |
| Fiber Deep #005 | 17080 | 3.23 | 16 | 4.95 |
| Fiber Deep #095 | 10625 | 2.01 | 10 | 4.97 |
| Fiber Rich #001 | 15323 | 2.90 | 18 | 6.20 |
| Fiber Rich #010 | 34070 | 6.45 | 36 | 5.58 |
| Fiber Rich #021 | 34357 | 6.51 | 43 | 6.61 |
| Fiber rich #036 | 17548 | 3.32 | 24 | 7.22 |
| | | | | |
| Fiber Deep Total | 80059 | 15.16 | 80 | 5.21 |
| Fiber Rich Total | 101298 | 19.19 | 121 | 6.40 |
| | | | | |
| Overall Total | 181357 | 34.35 | 201 | 5.80 |

*Does not include receiver / launch amplifier

Table 1.

Initial results of the Fiber Deep architecture study turned out to be slightly less expensive than similar plant sections designed using the Fiber Rich method. The simpler amplifiers and reduced cascades should also yield better long term operating cost and reliability.

The coaxial bus design of Fiber Deep has an optical node capable of bridger output levels and up to three line extenders in cascade. The existing trunk system is used to transport power and upstream signals within the service area. The reduced cascade allows for higher output levels at the same end of line distortions. These higher levels allowed use of fewer amplifiers per mile than required in the Fiber Rich architecture (Table 1.).

The optical network in Fiber Deep builds on the concept of having several fibers from the hub to the 500 home service area (six in Time Warner's case). Within that area, one or more field splits may be deployed to serve as many as four optical receivers. The fiber network is easily reconfigured for any combination of upstream transmitters and downstream splitters necessary to accommodate network traffic growth with very little additional cost and few customer interruptions.

Starting in mid 1994, Time Warner established a field test of the new architecture in Monroeville, PA. The system is a 330 mile plant in the suburbs of Pittsburgh, PA , with an average density of 145 homes passed per mile. Two hundred miles of the system are being upgraded using the Fiber Deep design. Fibers are split both at the hub and in the service area to provide optimum utilization of the optical power produced by today's laser transmitters.

## FIBER DEEP UPGRADE MAJOR COMPONENT CPM ANALYSIS

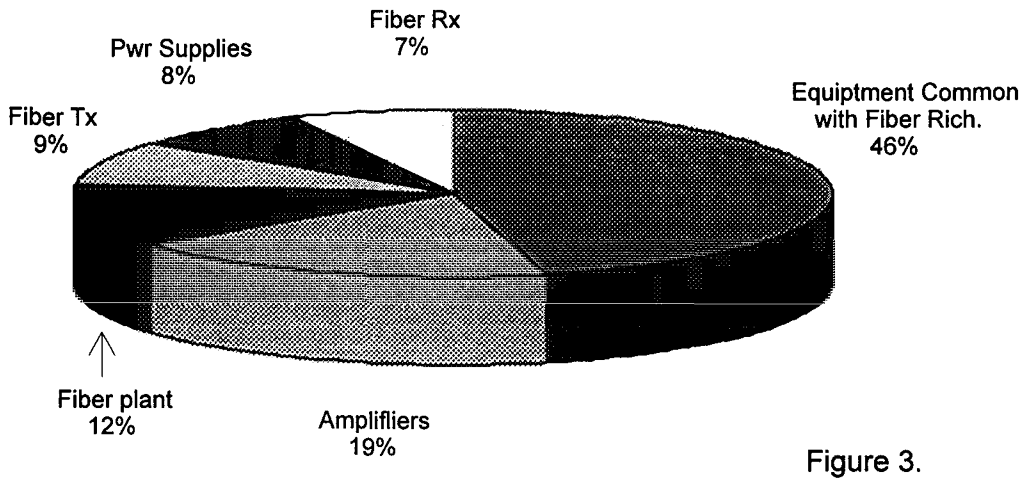


Figure 3.

Figure 3. shows the Fiber Deep architecture in percent of cost per mile of the major categories. The field numbers tracked our original paper study with an error rate of less than one percent. Besides having some operational and reliability efficiencies, the design also lends itself well to the implementation of HFC network powering of telephony. We feel that the concept should be cost effective down to the 70 to 80 hpm densities in plants where the trunk is 90 % or more aerial. The data used in this analysis represents 10% of the actual plant area. A more complete analysis should be available by mid 1995.

### Narrowcasting

As stated earlier, narrowcasting provides for real time, high speed, switched data services to be made available to our customers. Time Warner is currently reserving the top 200 MHz of all 750 MHz upgrades for these services. Our goal is to be able to deliver unique data streams to each service area, when the business dictates.

Today the most cost effective DFB laser transmitters are in the 8 to 10 mW output range. To utilize this much power, we must split the outputs to an average of four service areas, or 2000 homes passed per laser. The DFB laser manufacturers have yet to offer a cost equivalent lower power laser.

This may be the wrong way to look at solving this dilemma. True, the fiber architecture that is being incorporated calls for very short path budgets, but the objective is to be able to uniquely modulate each path. Maybe the answer is to split a high power laser source, then externally modulate the outputs to each service area. We may even be able to back up the laser source, solving some of the reliability issues surrounding the MTBF of DFB laser transmitters.

## Traditional Powering Architectures

The convergence of traditional CATV services with digital narrowcasting, and the advent of telephony over HFC have caused an increased focus on power delivery architectures. These architectures and the practice of powering design have not changed since the move from 30 to 60 Volts twenty years ago. The next several sections are devoted to a reexamination of these practices.

## Telephony Differences

Deployment of lifeline telephony service has placed new constraints on traditional powering architectures and design techniques. Time Warner Communication's current belief is that in order to maximize the market potential of telephony over coax, a prospective subscriber must perceive no difference in the nature, quality, and reliability of this service. These requirements have two important impacts on powering:

- In order to meet or exceed the traditional service availability offered by the local exchange carrier (LEC), a highly reliable powering architecture offering extended standby time (eight hours or longer) is required.
- Provision of telephone service must be accomplished without the need to place equipment inside or otherwise enter the subscriber's residence. This requires subscriber interface electronics to be powered from common (shared) network power sources.

## Increasing System Availability

System availability can be expressed in terms of subscriber-minutes per year of outage time: the outage time experienced by the *average* subscriber each year. In order to compete effectively with an entrenched LEC, the existing CATV network must be upgraded in a way that efficiently utilizes the embedded capital investment, and speeds the deployment of new services. These two requirements preclude the use of techniques which require a complete rebuild of the network. The fiber deep approach offers such efficiencies from an RF standpoint; The following concepts are being explored to accomplish similar advantages from a powering perspective.

## Extended Backup Options

In Time Warner's approach to the provision of telephony services, common power sources used in network powered architectures must offer at least 8 hours of standby time. Initial cost studies indicate that battery only standby power sources cannot compete on an operational basis with genset based sources due to the cost of battery replacements. Power sources with output ratings at or above traditional CATV capacities require battery banks similar in installed cost to genset based systems. Battery only systems offer deployment cost advantages only where small capacity units are required. Large gensets cost only moderately more than medium sized gensets. Telephony service deployment increases the total system power load per mile, calling for larger capacity power sources. These realities dictate that the number of power source locations be reduced to take advantage of the economies of larger genset based sources.

## Power Source Cascade Reduction

Power outages have historically been the most significant contributor to system outage time in the CATV industry. The standby power sources required by telephony services are expected to remain among the worst offenders in system availability calculations. As such, measures which reduce or eliminate subscriber service dependency on more than one power source (power source "cascades") have a disproportionately significant impact on improving system availability.

Single point node powering produces the maximum power system availability. In general, the 60 Volt design presently used in Time Warner's 500 home passed average nodes cannot achieve single point powering. Techniques under investigation to accomplish single point powering include use of 90 Volt power sources feeding equipment operating in a proportionally wider voltage range (to 40 Volts minimum), and the concept of node boundary determination based on power source service coverage areas.

Unfortunately, many existing CATV networks cannot be powered from a single location, even at the NESC codified limit of 90 Volts rms. This is due to network layouts that are "stringy" in nature, where loads are concentrated towards the ends of long feeders. This is often the case where natural and man made boundaries such as rivers and expressways prevent frequent interconnections. Low density networks are also difficult to single point power due to higher amounts of cable resistance for comparable loads. If node boundaries are determined by power source service area coverage in stringy and low density cases, optics cost would normally becomes prohibitive due to the small node sizes produced. The optical field splitting technique employed in the fiber deep approach addresses this concern from a forward RF perspective, and the presence of 4 or 5 unused fibers makes possible the deployment of additional return lasers for flexible sub-division of service area boundaries, to coincide with powering boundaries if necessary.

## Design Techniques for Reduced Power Source Cascades

Two techniques are currently available to the powering system designer to help provide the lowest power source cascades to the greatest number of subscribers (recall that system availability is defined as outage time experienced by the *average* customer).

The first technique is to serve the greatest possible number of subscribers from a single point location, with the remaining voltage starvation cases addressed using an "extender" power supply. This supply would generally be a pole mounted low capacity unit offering 8 hours of standby time with batteries alone. Because of the limited number of subscribers served by the extender supply, the system availability calculation is not significantly impacted.

The second technique is to segment the node into "cells", taking care to place power source service area boundaries at the optical receiver location. Some type of switching methodology is then provisioned at the receiver whereby power failure in one cell will automatically switch the receiver power source to another cell. This technique maintains a power source cascade of one, with only a small reliability hit due to the automatic redundant source switch. This technique can be used to further subdivide a fiber deep service area below the level attainable through deployment of additional return lasers.

Another technique under investigation is the use of an autotransformer to step up voltage in order to address voltage starvation at the ends of long feeders. This technique is not available to designers because the required transformer and power removal/insertion device do not exist. The same technique could be used in reverse to limit power passing tap voltages in drop cables to 60 volts for power delivery to network electronics in locations subject to NEC regulations (inside buildings, etc.), while allowing use of 90 Volt transmission in the express portion of the network.

## Dynamic Power Loads

The techniques described above are complex enough for the static loads of existing networks. Network powered telephony introduces two additional variables which vastly complicate the task of designing efficient powering architectures. These two variables are equipment usage patterns (traffic) and equipment deployment patterns (penetration rates and distribution).

## Traffic

The leading proposals for the provision of residential telephony in existing coaxial networks employ an electronics package attached to the outside of the residence. This package is designed to closely resemble, and be located near the existing telephone network interface. This package, commonly referred to at the NIU, is being designed to minimize power consumption. As such, the NIU draws less power in the idle state than when subscriber telephones are off-hook or ringing. In the large sample sizes served by LEC switches, this traffic is well defined, and equipment can be provisioned for much less than 100% usage with a very low probability of call blocking. In the sample sizes typically served by a CATV power source location, this becomes a dangerous practice. If call volume is allowed to exceed the capacity of a given power architecture, then calls are not merely blocked; a shared outage is experienced by all subscribers in the affected area, including CATV customers!

The alternatives are to employ a software blocking algorithm to prevent such an activity, or to provision powering architectures capable of 100% loading. The software approach requires a complex software model of network topology, and is likely to produce blocking in excess of LEC standards if not accompanied by some degree of purposeful powering overdesign. The latter approach may be attainable at reasonable cost if deployed in conjunction with a solution to the more serious problem of penetration rates and distributions.

## Penetration

Telephony penetration in the early years of deployment is expected to be relatively low, and even long term penetration is expected to be well below 100%. At first blush, this seems to be a ready opportunity for significant cost savings. However, two problems must be addressed before this cost savings can be realized.

## Penetration Distribution within a Power Source Service Area

The first problem is caused by the uneven "lumpy" distribution of customers at a given penetration rate. The treelike nature of coaxial networks increases the likelihood of problems caused by concentrations of customers appearing in worst case locations such as the end of long feeders. These concentrations, in conjunction with the aforementioned traffic effects, have the potential to produce localized temporary outages due to voltage starvation conditions. These voltage starvation effects are best addressed by testing the powering design computer model under various loads, such as statistical off hook in conjunction with 100% penetration, 100% ringing under 100% penetration, etc. The results of this modeling are then used to call to the designer's attention plant segments which present unusual or poorly balanced powering designs, thus increasing the margin of safety at sub-100% penetration power system deployment levels.

## Penetration Distribution between Power Source Service Areas

Perhaps the most difficult design problem is posed by the relative penetration success of one power source service area with respect to another. The small sample sizes offered by these service areas increase the probability of excess penetration in one area with respect to another. The nature of demographic effects in such small sample sizes is unknown, as is the amount of aberration from a normal statistical distribution due to factors such as the "word of mouth" popularity expected as customers try out the service in individual neighborhoods. Once the voltage starvation effects of lumpy distribution and traffic are addressed using the load modeling technique, then modular power sources can be deployed such that installed capacity can be increased on an as-needed basis without need to relocate these typically large and difficult to site power sources.

## Conclusions

A number of important issues still exist surrounding CATV industry entry into the new services proposed for the information superhighway. There are answers for the majority of the technical issues, although many of them are currently very expensive. In the early deployment stages, many of the technical issues such as network powering and narrowcasting capacity must be over engineered to avoid the risk of market failures due to problems with system availability. Initial system deployment must be carefully monitored for margins, and the knowledge gained used to increase the efficiency of subsequent designs This approach will enable CATV systems to be competitive on a cost basis, and will allow the CATV industry to deploy a service equal to or better in nature, quality, and reliability compared to the competition.

The marketing side is probably even more hazy. There are many projects in progress to help clear the way (Time Warner's Full Service Network in Orlando). We believe the key to success is to keep an open mind and use all the tools available to build a flexible network–one which can be changed without taking a major detour on the information superhighway.

(1) Annual down-time analysis of selected broadband networks, by: Walt Strode, Director, Quality assurance, Phillips Broadband Networks, Inc.

## Customer Focused Service Process Improvement

Nicholas Worth

### Abstract

*Cable operators need to improve productivity and service quality to compete successfully in the future. Customer focused service process improvement and reengineering have helped "breakthrough" service firms achieve both goals, as evidenced by examples in the paper.*

### Introduction

Cable television operators need to improve productivity to restore cash flows lost to rate regulation, which is merely a placeholder for rapidly developing competition. To avail themselves of new opportunities in telecommunications and fend off competitors, MSOs must improve brand images and loyalty with consumers. But brand image and loyalty must be built on a foundation of genuine service quality. Further, improvements in quality must come at a time when capital requirements are at historic highs.

How does a service business improve operating efficiencies to restore cash flows necessary for new investment, while simultaneously increasing service quality? Fortunately, a number of U.S. service firms have shown us how to achieve these goals through customer focused service process improvement and reengineering. Other firms have shown us how to fail through classical cost cutting measures which did not take customer needs into account.

If you do not accept the imperatives of increased quality, efficiency and customer responsiveness, consider the following.

### Competition Brings Change

As recently as the late 1970's, Swiss watch makers controlled two-thirds of the wristwatch market. Within three years the Swiss watch making industry was devastated by the introduction of the quartz (electronic) watch. Employment fell by 75%. Yet, the Swiss Watch Federation's Research Center actually developed the quartz watch. But, it did not tick and to the product (internally) focused Swiss, it was not a watch. Had they been customer (externally) focused, the Swiss would have seen immediately the consumer benefits of precision, low cost, abundant features and freedom from rewinding. And so, old, established brands such as Bulova were replaced with new brands such as Seiko whose firms were more efficient and customer focused.

Are there any such threats on the horizon for cable operators? Direct TV/USSB have begun filling an unserved niche for rural service, expanded programming choice and clearer pictures. What if satellite receiver costs follow the experience curves of VCRs? How will you feel about competing with Direct TV after Sony and NEC have entered the market and prices have fallen to $399?

But, there are many other competitors who want a piece of the home entertainment and information delivery pie.

| | |
|---|---|
| American Wireless | GTE |
| Ameritech | Nynex |
| AT&T | Pac Tel |
| Bell Atlantic | People's Choice |
| Bell South | SW Bell |
| Blockbuster | US West |

The list is long and distinguished.

**Cable's Current Position**

At present, cable operators have very high share of the market for basic and cable programming services, which are maturing products. Normally, high share in a mature market for which demand is strong, leads to high profitability (the cash cow shown in the figure). Cash cows normally provide funds to fix problem children such as pay-per-view or provide funds for potential stars such as advertising sales or perhaps broadband telecommunications.

However, profitability of basic and cable programming services is low.

The FCC's punitive rate regulation artificially pushed

| Growth | H (Profitability) | L (Profitability) |
|---|---|---|
| H | Star | Problem |
| L | Cash Cow | Dog |

Profitability (share)

**BCG Portfolio Model for Product Analysis**

these products into the "dog" quadrant. So, how do cable MSOs make their core products profitable again?

**Tools Available For Productivity Improvement.**

Each of the following strategies can, if properly applied, improve a firm's productivity ratio of output ($ sales) to inputs (labor, capital). Two of the strategies, if well implemented, can simultaneously lead to service quality improvements.

**Restructuring** is a euphemism often applied to the process of radically reducing an organization's operating expenses. Managers in competitive environments should always be alert for opportunities to reduce or prevent expense for projects which add no value for customers. But, cable operators have always done a credible job of expense control. As reported in a recent issue of the *Wall Street Journal,* IBM auctioned off valuable art works from its headquarters. This news is probably difficult to understand for most cable operators who have traditionally avoided this type of extravagance.

When is classical cost cutting counterproductive? Sears, in the late 1980s, replaced 70% of its full-time sales persons with part-timers to reduce salary and benefit expense. Customers fled, apparently frustrated by the resulting poor service, negating any savings. Profit margins slid from 4.9% to 1.2%. Sears neglected their customers' needs in their search for cost savings.

Many believe that **capital investment**, e.g. in information technology, will simultaneously improve service quality and productivity. Since 1980, U.S. service firms have invested more than $400 billion in information technology with little measurable increase in productivity. Why? According to Dartmouth College's James Bryan Quinn, many firms merely automated inefficient processes already in place. A few firms, such as L.L. Bean, **reengineered** their service processes around their customer's needs before automating and reaped huge rewards in increased sales. Hopefully, cable's investments in fiber to improve reliability, capacity and picture quality will pay similar dividends by filling customer needs.

**Reengineering** has unfortunately become, of late, a buzzword used to cover a multitude of strategic sins. Properly done, however, service process reengineering has yielded huge gains for firms such as Taco Bell. Taco Bell's customer surveys in the late 1980's showed that customers valued the food, the cleanliness and appearance of the restaurant, the service and nothing else. Taco Bell redesigned the service process around its customer contact persons and increased their training. They simplified back room operations by outsourcing food preparation to suppliers, freeing customer contact people to interact more with customers. By investing in customer contact persons, Taco Bell was able to eliminate two layers of middle management who added no value, reducing payroll expense by 50%, while growing sales and profits dramatically. But Taco Bell is a division of Pepsico, a firm with vast experience in process reengineering.

For most firms, **continuous process improvement** is the safest starting point. But what exactly is continuous process improvement? The following example from manufacturing will help explain the concept.

Nashua Corp. was a manufacturer of coated paper in the early 1980s when CEO Bill Conway invited W. Edwards Deming to teach continuous improvement principles to his managers. Because customers were dissatisfied with the marks made by writing pens if coating thickness were too thin, the company inspected coating thickness on each production run and rejected paper which did not meet minimum thickness specifications. The tendency was therefore to err on the thick side, which was also wasteful. Using process control charts, Nashua engineers were able to identify and eliminate the causes of variation produced by the coating machine. The results are depicted in a process control chart, Figure 1, at the end of the paper.

Once the process was in statistical control, 100% inspection and rejection of defects was no longer necessary and the Nashua was actually able to reduce coating thickness, saving $millions per year. Why not avoid a lot of trouble and buy a new coating machine, you might ask? The machine was new.

Nashua Corp. used these continuous process improvement techniques in the manufacture of computer hard disk surfaces to achieve dominant market position and drive other North American manufacturers out of the business.

**Service Industry Examples**

An outstanding example of continuous service process improvement was provided by Midway Airlines. Midway transformed themselves from a commodity based competitor who survived on the overflow from other airlines to a quality leader who commanded premium prices. Midway's customer research determined that on-time performance was the number on customer requirement. Midway used management led, cross functional teams of employees to identify and measure the root causes of delays in departures. As shown in the Pareto chart (Fig. 2 following the paper), four causes were responsible for 88 % of the delays. Midway's team developed solutions for each of the causes and improved on-time performance from 65% to 95%.

Continuous process improvement may seem like a lot of work for very little gain. However, the following example illustrates the leverage which can result from a 1% improvement in market share and expense reduction.

<u>Before Improvements</u>

| | |
|---|---:|
| Sales | $1,000,000 |
| Expenses | <650,000> |
| EBITDA | $350,000 |
| Dep. & Amort. | <150,000> |
| Interest expense | <150,000> |
| Earnings (bef./tax) | $50,000 |

<u>After Improvements</u>

| | |
|---|---:|
| Sales | $1,010,000 |
| Expenses | <643,500> |
| EBITDA | $366,500 |
| Dep. & Amort. | <148,500> |
| Interest expense | <148,500> |
| Earnings (bef./tax) | $69,500 |

The result is a 39% increase in profits for a 1% improvement in sales and expenses - talk about leverage!

Federal Express used continuous service process improvement to maintain its leadership in overnight small parcel delivery. The firm's pioneering work in developing a balanced measurement system (people - service - profits) is a model for all service firms to follow. Federal Express learned that if you do not measure it, you will probably not improve it.

**Cable TV Examples**

TeleCable Corporation sought to follow Federal Express' lead in developing its Service Quality Index which measured performance in five areas that customer research showed were critical.

| Respond to customers: | Weight |
|---|---|
| Quickly | 20% |
| Conveniently | 20% |
| Competently | 20% |
| Courteously | 15% |
| Reliable delivery: | 25% |

For example, TeleCable's reliability objective was 99.99% and the firm measured performance against that goal.

Following are two examples of service process improvements achieved by management sponsored employee teams in TeleCable systems.

The advertising sales staff in Arlington, Texas was concerned about wasted inventory ("make goods") and client dissatisfaction with missed commercials. An employee team examined the entire commercial insertion process, identifying causes of failure and measuring their frequency. The first breakthrough was achieved by adjusting studio hours (no staff hours were added) to assure that a trained person was in attendance during weekends when catastrophic failures were occurring. Other improvements included sample testing schedules after downloading and the replacement of one old inserter (the only capital expenditure). Results were dramatic as shown by the run chart (Figure 3 following the paper). Clients were more satisfied and this increase ultimately led to higher sales. Wasteful "make goods" were largely eliminated.

Ron Moore, General Manager of the cable system serving Cleveland, Tennessee, was dissatisfied with the quantity of trouble calls and the quality of new installations. He formed an employee team to seek improvements. The team examined the installation process and concluded that the scheduling system was in effect serving as a quota which encouraged only quantity and not quality. The team suggested that if they were allowed to perform the installation correctly the first time, including measuring signal and leakage levels and educating customers about cable TV service, the small amount of extra time spent would be more than offset by reductions in trouble calls. They were right and the Cleveland system became a test bed for an engineering department developed concept called the "lifetime installation". Trouble calls associated with new installations fell from an average of 100 per month to 50, then 33, then 20 and finally 15 where they have remained!

These and many other process improvements throughout TeleCable combined to offset the financial effect of the FCC rate rollbacks and increased customer loyalty as evidenced by strong penetrations and improved customer survey results.

**Service Breakthroughs**

In their book, *Service*

*Breakthroughs,* authors Heskett, Sasser and Hart determined two things which distinguished "breakthrough" service firms from "merely good" firms. The managers of the merely good firms viewed productivity and service quality as trade-offs to managed, whereas managers of breakthrough firms viewed the two as part of a continuous, reinforcing cycle of satisfied customers and lower costs.

Managers of merely good firms valued each customer in terms of the pending transaction (e.g., a $25 purchase), whereas breakthrough managers were mindful of the lifetime value of a satisfied, repeat customer - estimated by Club Med to be $4,000.

**Summary**

Customer focused service process improvement provides a way to simultaneously improve quality and lower costs. Success requires a deep commitment from the top of an organization all the way to the front line. Once a firm has acquired skills in process improvement, that firm will be well prepared for process reengineering opportunities which may yield even larger gains. In each success story the author has discovered, the search for improvement began with the customer - the only person who can define quality.

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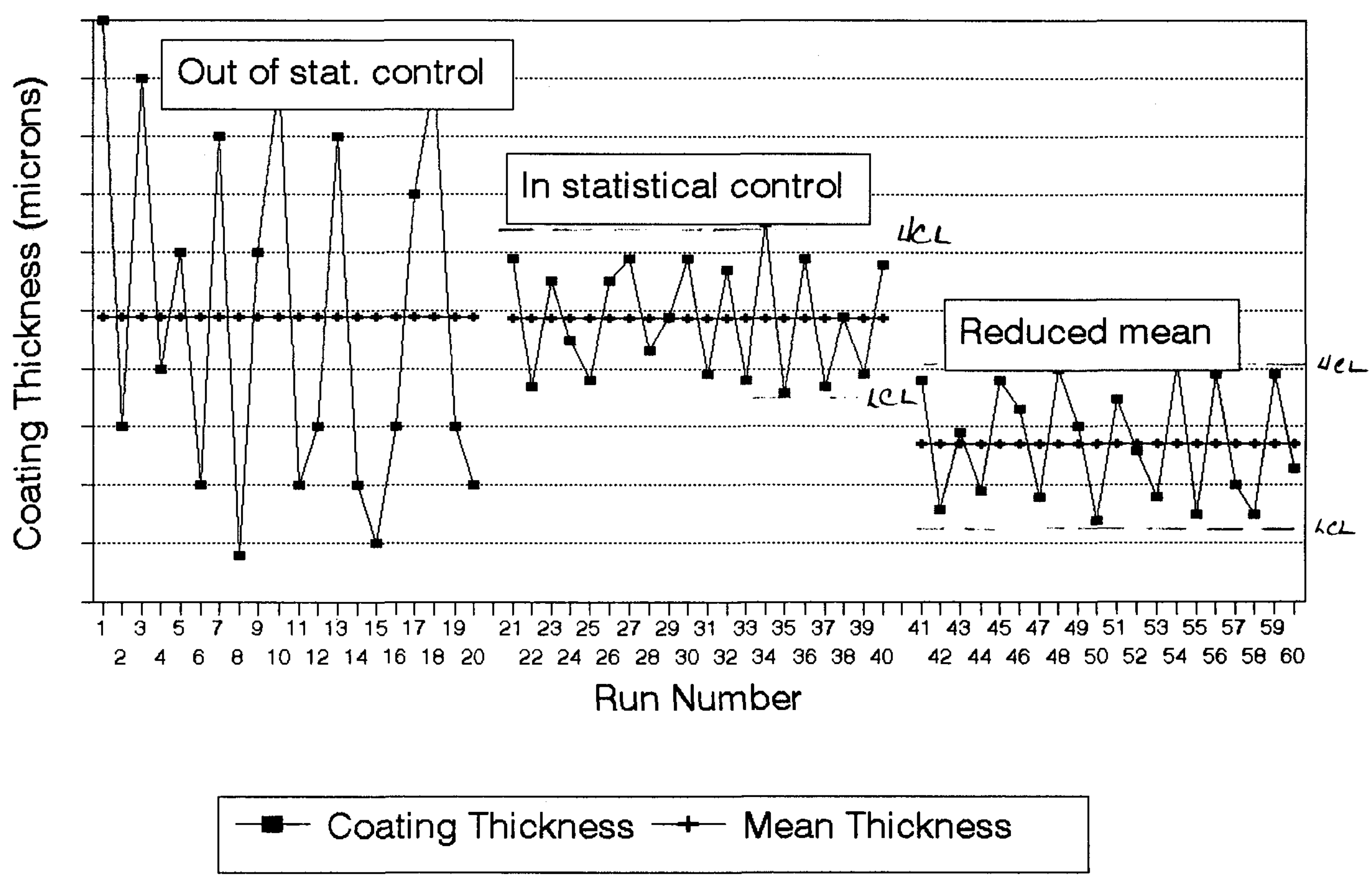
"The Service Driven Company", *Harvard Business Review,* Sept.-Oct., 1991.

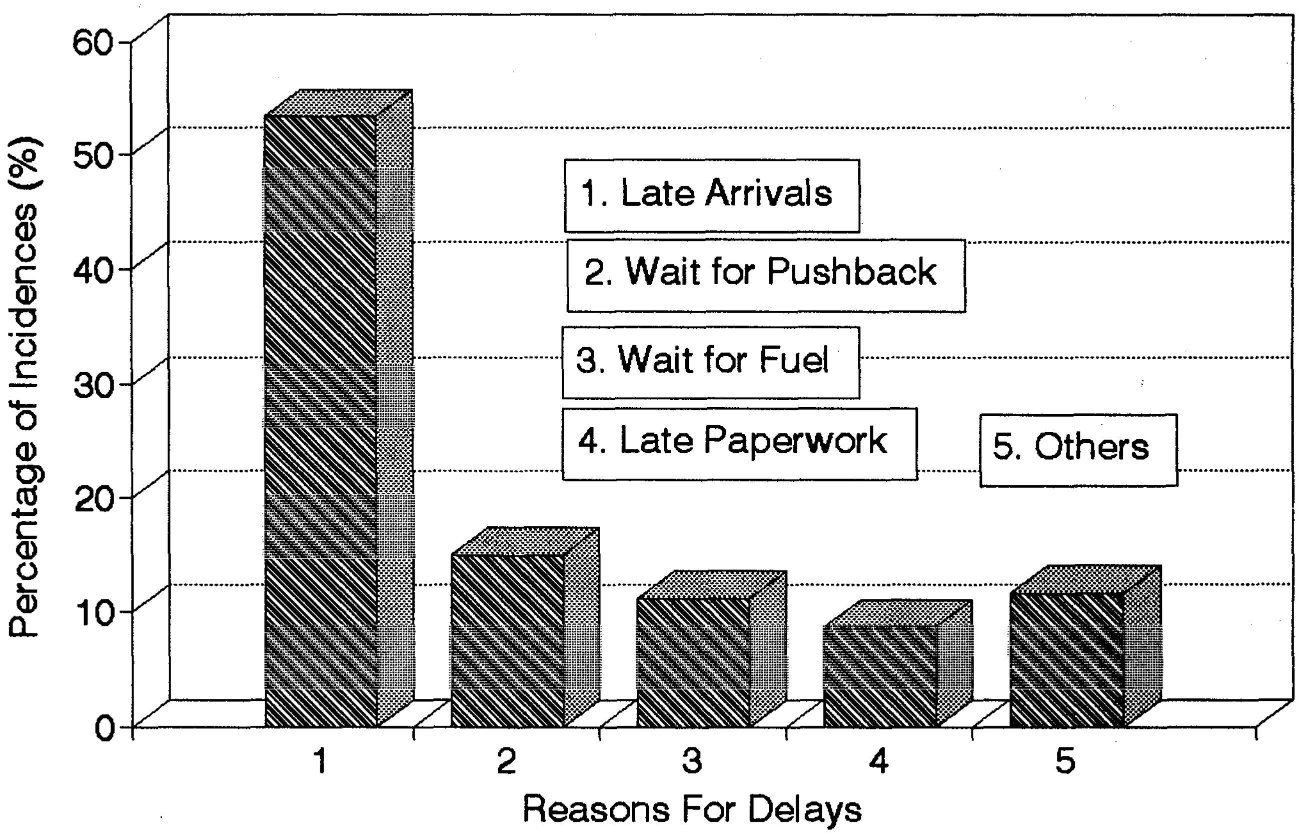
# Fig. 1 - Nashua Corporation

## Paper Coating Thickness vs. Run #



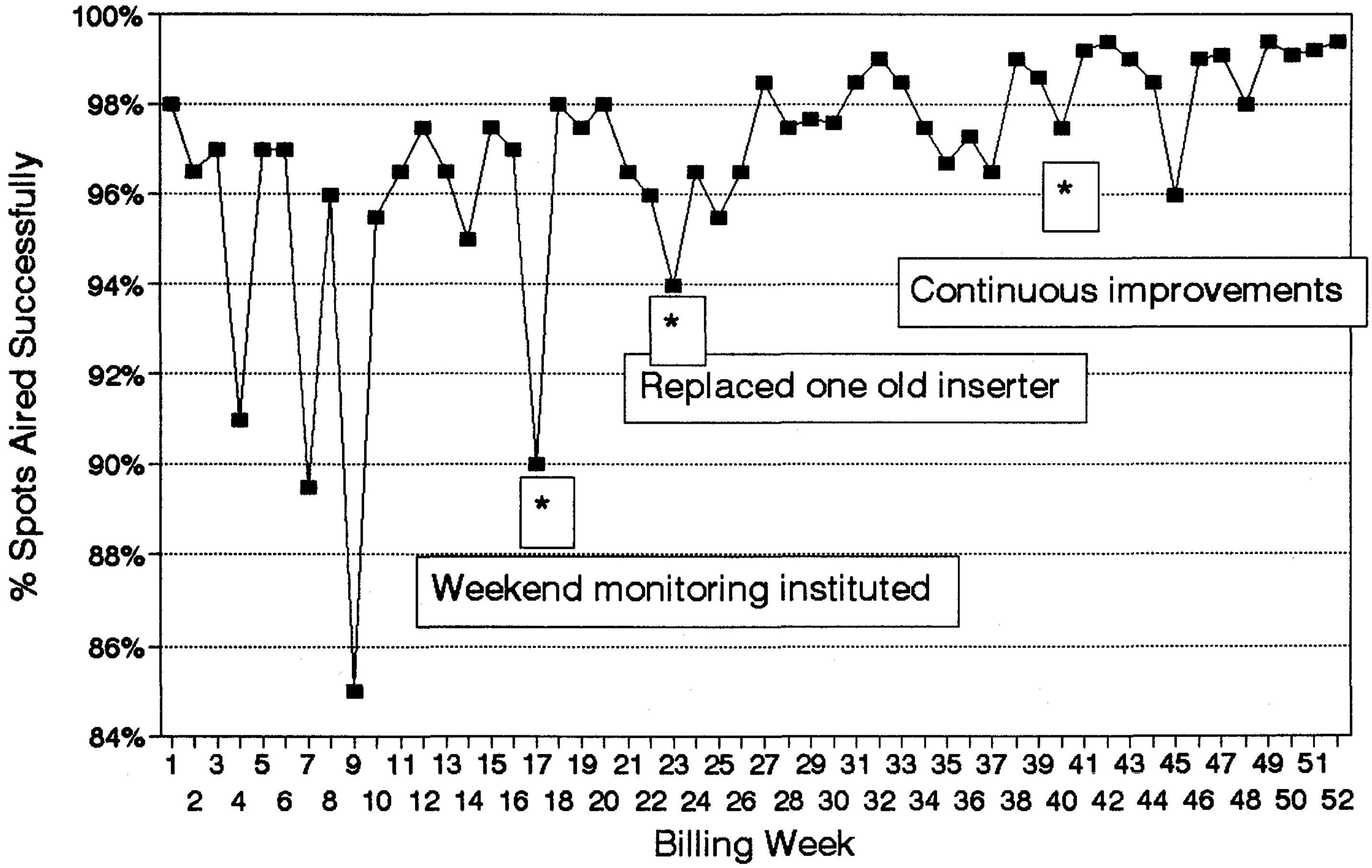
# Fig. 2 - Midway Air Pareto Chart

## Reasons for Flight Departure Delays



# Fig. 3 - Arlington Run Chart

% of Spots Aired Successfully vs. Week



# Digital Video Transmission System with Pilot Aided C-OFDM

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## ABSTRACT

*A digital video transmission system is proposed that uses a pilot-aided Coded Orthogonal Frequency Division Multiplexing (COFDM) scheme with multiple carriers which are individually modulated using QAM. Multiple MPEG video channels are supported within a 6MHz band. OFDM is known for its multipath mitigation effect.*

*The technique is described from an implementation point of view, addressing such issues as equalization of channel characteristics, AFC, AGC and timing synchronization, which are critical for realizing the system. Trellis decoding is also discussed with respect to equalizing the channel characteristics, which improves the bit error rate performance. The performance of the system are shown by the computer simulation.*

## 1. INTRODUCTION

Digital video transmission schemes using QAM and VSB have been proposed recently[1][2].

CATV transmission line has reflections on the channel. The performance is degraded by channel impairments such as reflections and impedance mismatches. OFDM is a robust technology for overcoming the reflections. We propose a digital transmission system using OFDM that achieves good bit error performance using Trellis-Coded Modulation and Reed Solomon code.

## 2. OUTLINE of TRANSMISSION SYSTEM

Figure 1 shows the outline of the transmitter of the proposed system. Four video programs with MPEG 2 coded digital TV source are accommodated. These video source data are multiplexed into a single stream in an I/F board.

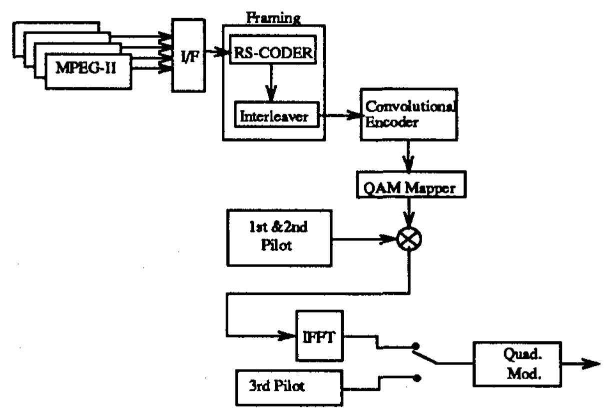


Figure .1 Configuration of OFDM System

The multiplexed data is fed into a Reed Solomon encoder. We use an RS(208,188) code with 10 symbol error correction capability. The output of the Reed Solomon encoder is interleaved symbol-by-symbol of the RS code words. The output data of the interleaver is fed into a convolutional encoder. The convolutional encoder processes the input symbols five-bit at a time. Encoded 6-bit vectors are mapped into QAM vectors. A single OFDM symbol contains 512 carriers, with 10 KHz carrier spacing, each of which is modulated using

64-QAM. 512 symbols of QAM for 512 carriers of each symbols are transferred to the time domain using an IFFT, resulting in a group modulating 512 QAM carriers.

A 1024 complex data IFFT which generates the baseband OFDM signal in the time domain. The time domain OFDM baseband complex data is converted into an IF signal by a quadrature modulator realized by a digital interpolation filter and multiplexing.

## 3. FRAME FORMAT

The frame structure is shown in Figure 2. The frame consists of three pilot symbols as preamble, and 4 data frames each of which has 52 OFDM symbols. The pilot symbols are inserted every 21.1 msec.

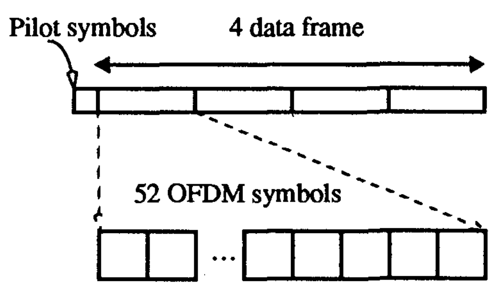


Figure .2 Frame structure

### 3.1 RS Error Correcting Code and Interleaving

We use a concatenation error correcting code with an RS code and Trellis-Coded Modulation for error control. The RS code is used for channel coding because it has good performance in burst error conditions, providing a reliable channel.

We choose an RS (208,188) code which can correct upto 10 symbols. Each symbol contains 8bits. The interleaving matrix design takes into consideration of the RS code length and the number of carriers in one OFDM symbol.

Figure 3 shows the interleaving matrix. The data frame consists of the interleaving matrix of 208 bytes by 80 RS code words, which corresponds to 128 carriers space interleaving depth. The interleaving spreads erroneous symbols up to 128 carriers consecutive errors. A whole OFDM symbol corrupted, only four symbols of any single Reed Solomon codeword may be corrupted, which can be completely corrected.

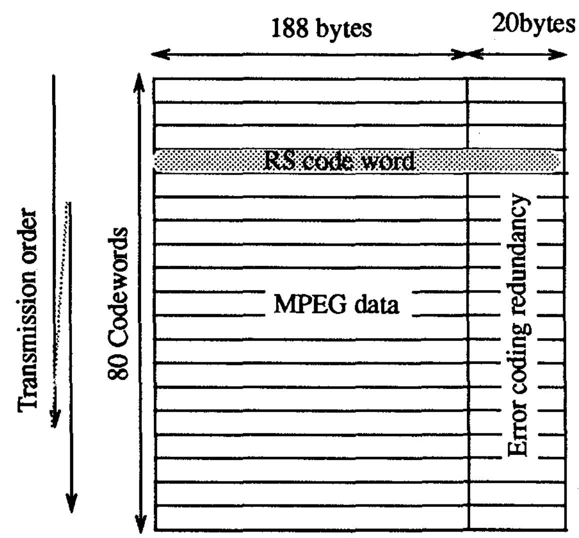


Figure .3 Interleaving matrix

### 3.2 Pilot Signal

We use pilot symbols to realize channel equalization, timing synchronization and frequency offset correction. Three pilot symbols are followed by 4 data frames. Each data frame is shown in Figure 3, which is the interleaving frame. The first pilot symbol is a null signal period and the second is a sweep signal and the third pilot symbol is an ASK signal which is modulated by a PN sequence.

The first symbol can be used for coarse

timing detection for the data frame, however symbol timing precision is realized by the third pilot symbol.

The second pilot symbol can be used to measure the channel characteristics. All components in a frequency band (5.12MHz) of the second pilot has the same level so that we can detect the channel characteristics with equal precision in a bandwidth. The details of the second pilot is shown in the AFC explanation below.

The third pilot symbol is an Amplitude Shift Keying, which can be used for both timing synchronization and frequency control of received signal.

## 4. TIMING SYNCHRONIZATION

The carrier in the third pilot is modulated by a PN code of length of $2^7$-1, which is generated by 7 bit linear feed back registers. At the receiver, the envelope detection is used for demodulation of ASK signals modulated by the PN code.

The PN code has a very sharp auto-correlation over additive white gaussian noise of the channel. The PN code of length 127 gives approximately 21dB of code separation and hence can be detected in the channel with C/N of 21dB.

The output of the correlator gives the position of the preamble of the data frame, and the timing precision is the reciprocal of the clock rate of the PN code. We use 1.28MHz of the clock of the PN code, and the PN peak point can be detected by the precision of 0.78μsec.

The PN code is detected by a correlator which is used as a synchronization word detector. We use the synchronization word protection to achieve quick and secure acquisition, and to release a lock in the case of a false lock. For the backward sync-word protection, a sync-word with $N$-bit errors is allowed to set the lock. Once synchronization is achieved, we allow a sync-word with $M$-bit errors. This realizes the synchronization with data frame.

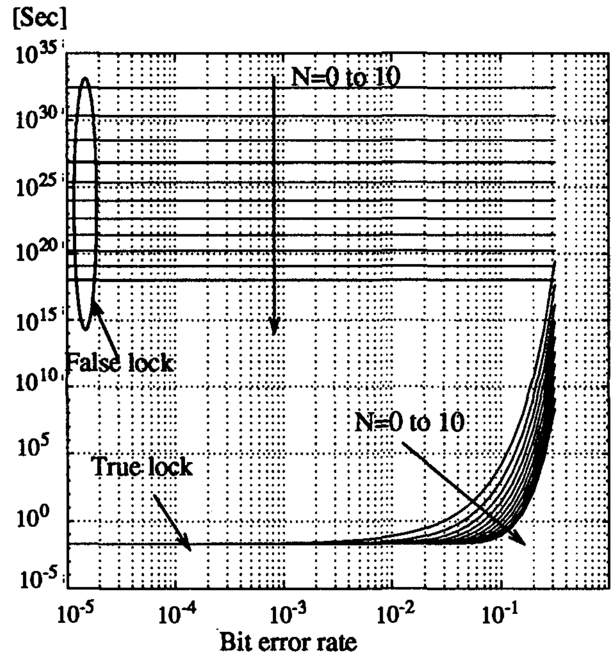


Figure .4 Lock up time for PN code synchronization

$N$ and $M$ can be determined by computer simulation. Figure 4 shows the time in seconds for the system to lock with $N$=0 to 10. When $N$ is zero, no error is allowed, it takes approximately 27hours ($10^5$sec) to lock at a BER of $10^{-1}$ and 21msec to get the lock state. On the other hand a false lock happens every $10^{18}$ sec. with $N$=10, and $10^{31}$ sec. with $N$=0. We determine that if $N$ is 10, a lock time is 21.1msec and a false lock occurs every $10^{18}$ second. Figure 5 shows the time in msec. for the system to unlock with $M$=10 to 20. When $M$ is 20, a lock out occurs every $10^{15}$sec. from a true lock, and it takes $10^{-6}$ sec. to unlock.

Another benefit of ASK signals, the received signal can be demodulated even with frequency offset. The advantage of this method is that the timing synchronization can be achieved with some frequency offset of the received signal. Moreover the ASK signal can provide the frequency information for the selection of the PLL-synthesizer frequency step of a tuner at the first stage of receiving.

Thus, channel selection for the tuner and symbol timing acquisition can be obtained by using the third pilot symbol which is the 2-Ary ASK signals.

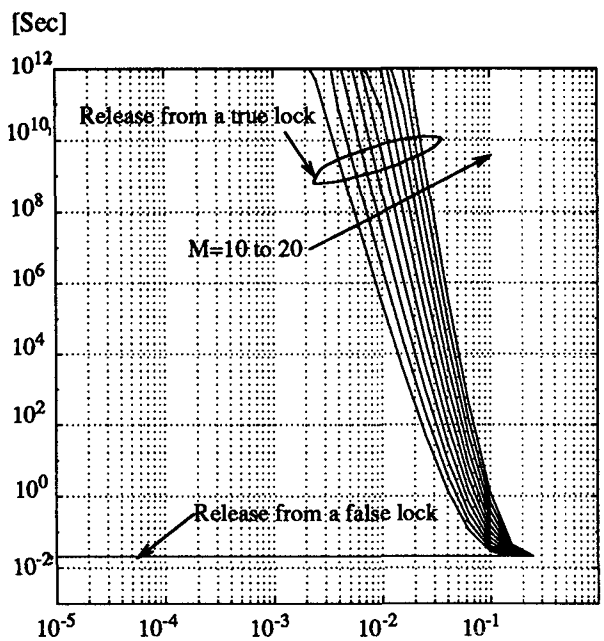


Figure .5 Release time of PN synchronization

# 5. DIFFERENTIAL DECODING and AUTOMATIC FREQUENCY CONTROL

We can use the second pilot for differential decoding and automatic frequency control. The second pilot symbol can be designed for differential decoding and frequency detection and gain control.

## 5.1 Description of the Second Pilot

The second pilot should be equal level for all carriers, which is an appropriate symbol for calculating the channel characteristics, in terms of a well-balanced dynamic range of equalization.

The pilot symbol should have a constant level in the time domain so that we can get a level information for AGC.

Considering the above features, we use the sweep signal shown in the Eq.-(1).

$$P(t) = exp(jwt)$$

$$= exp\left(j\pi\frac{F_s}{T_s}t^2\right) \quad (1)$$

,where $Ts$ is the symbol duration and $Fs$ is the nyquist bandwidth of OFDM symbols. The equation indicates that the pilot symbol has a constant level for AGC. The pilot symbol in a frequency domain can be written in this form;

$$P(f) = exp\left(j\pi\frac{T_s}{F_s}f^2\right). \quad (2)$$

We can see that the absolute values of each carrier's component are identical to *1.0*.

## 5.2 Differential Encoding and Detection

Suppose *S(f)* is the transmitted OFDM symbol and *H(f)* describes the channel characteristics. We can encode OFDM symbols differentially with the pilot symbol. The received symbol can be written in the following form;

$$R(f) = H(f)P(f)S(f). \quad (3)$$

and the received pilot symbol obtains the same channel characteristics and written in the form;

$$Rp(f) = H(f)P(f). \quad (4)$$

As in Figure 6, we can cancel the channel characteristics at the receiver by just dividing symbols by the received pilot symbol.

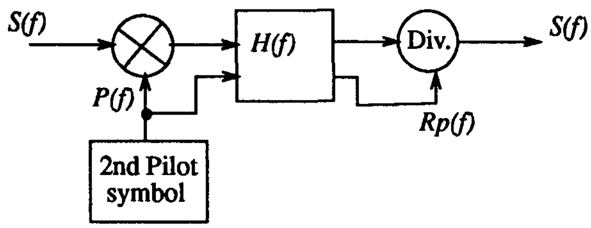


Figure .6 Differential Encoding and detection

### 5.3 Automatic Frequency Control

The receiver typically uses a PLL synthesizer for the tuner, where the tuner output frequency can be controlled by the synthesizer at 250 KHz step. The output signal of a tuner is converted into a complex baseband OFDM signal. Using an NCO we can correct the frequency offset. The frequency offset is obtained by manipulation of the second pilot symbol of Eq.-(1).

The phase is detected as

$$\Phi(t) = \pi\frac{Fs}{Ts}t^2 + 2\pi\varepsilon t \qquad (5)$$

,where $\varepsilon$ is frequency offset. Now the receiver holds the phase information of the second pilot symbol as an reference written as below:

$$\Phi_{ref} = \pi\frac{Fs}{Ts}t^2. \qquad (6)$$

We subtract Eq.-(6) from Eq.-(5) and take the derivative of the result and obtain the linear function of the frequency offset.

$$\frac{d}{dt}\Delta\Phi = 2\pi\varepsilon \qquad (7)$$

Using Eq.-(7) we can calculate the frequency offset and forward it to the NCO, forming an AFC loop. The additive white gaussian noise can be cancelled by averaging 8 frames of data.

Symbol error rate performance of the receiver was simulated and shown in Figure 7. We confirmed that the frequency offset is determined, even for a lower C/N. For the channel distortion, we could use the linear estimation with minimum mean-square.

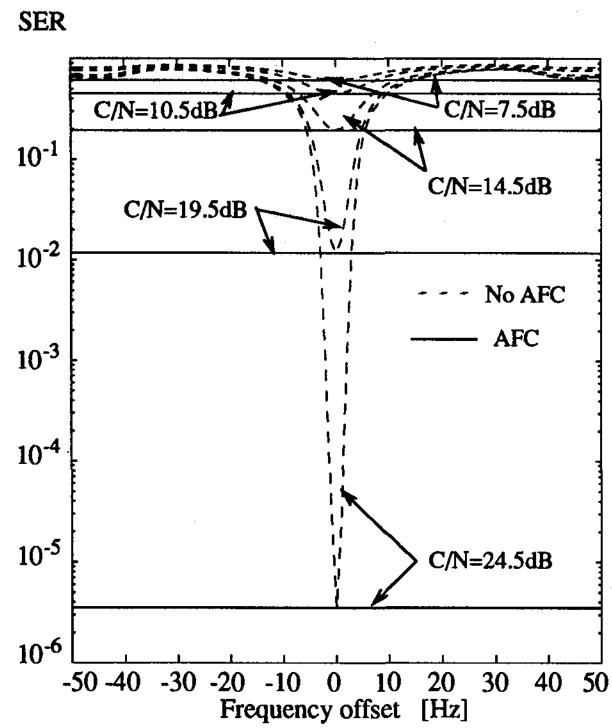


Figure .7 Symbol error rate versus frequency offset

## 6. BIT ERROR RATE PERFORMANCE of SYSTEM

### 6.1 Channel Model

The CATV channel is modeled from the view point of the physical channel architecture. Impedance mismatches of the trunk amplifiers generate the signal fluctuation within a bandwidth. In the case of OFDM signal, multiple carriers receives the signal fluctuations in the band. This can be caused by the delay version of the signals with different delay times. Usually we can use the model [2] considered for a cable architecture. For simplicity, however, we can use the model of a Rician channel, shown in Figure 8. The Rician model is applied to all carriers in a given bandwidth for each OFDM symbol. This model can be used to simulate the fluctuation of carriers in a given bandwidth due to reflections and losses of the channel.

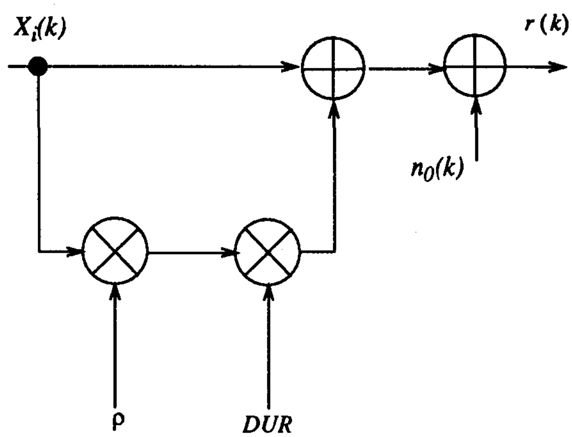


Figure .8 The channel model using Rician variables

This model of channel conditions is more severe than the condition of a real channel.

### 6.2 Differential Demodulation with the Second Pilot Symbol

As we discussed above, the second pilot symbol has equal level components of each OFDM carriers in a bandwidth. The pilot symbol receives the same distortion in a channel as the OFDM symbols, so each OFDM carrier multiplexed by the pilot symbol in the frequency domain, can be demodulated through division by the received pilot symbol which also contains the channel distortion. As a result we can cancel the distortion in the frequency domain by simple division by the pilot symbol. The performance of the bit error rate for coherent demodulation and differential demodulation of 64QAM/COFDM are compared in Figure 9 and Figure 10.

Figure 9 shows the bit error rate performance of coherent detection which is identical to the case of a single carrier 64QAM and 8VSB without equalization, at 0dB,12dB and 24dB of D/U. Figure 10 shows the bit error rate performance of a differential detector for 64QAM/COFDM with pilot symbols at 0dB,12dB and 24dB of D/U. We can notice the improvement of bit error rate performance by the channel characteristics equalization, which is differentially demodulated using the pilot symbol.

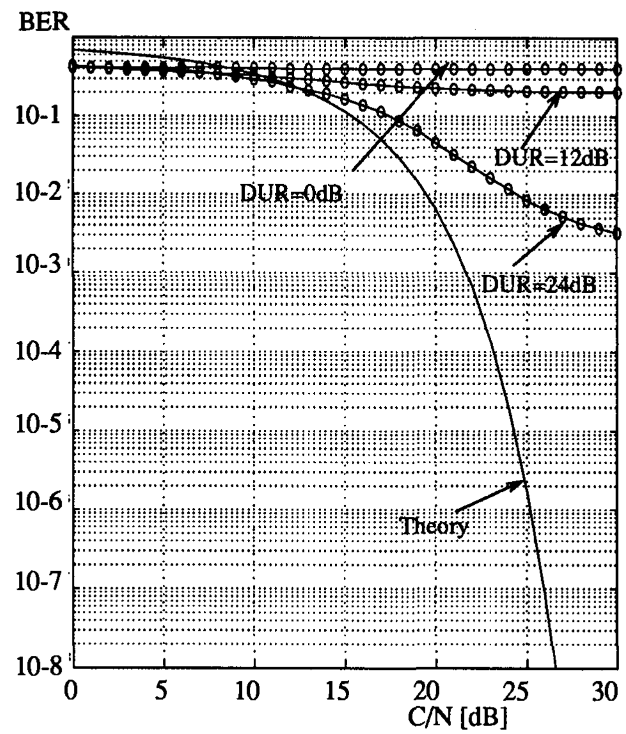


Figure .9 Bit error rate for 64QAM/OFDM by coherent detection at 0, 12, 24dB of DUR

### 6.3 Trellis Coded Modulation with Pilot Symbol

Trellis-Coded Modulation can provide high coding gain. We can further improve the coding

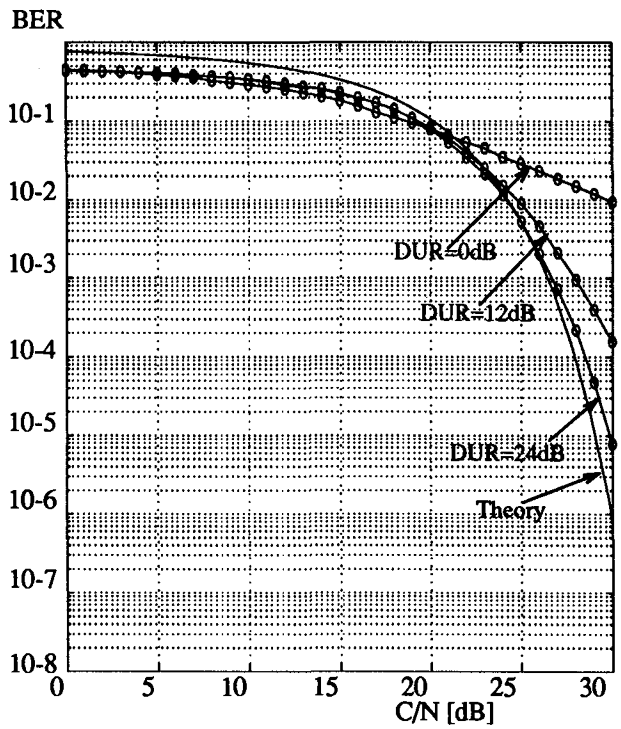

Figure .10 Bit error rate for 64QAM/COFDM with differential decoding by pilot symbols

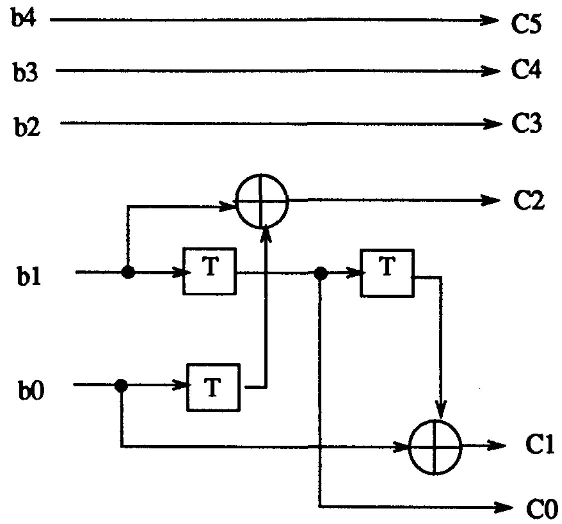

Figure .11 The convolutional encoder for 64 QAM/COFDM

gain of the system by using an RS code as an outer code. Trellis-Coded Modulation can be applied to correct the channel impairments, such as reflection and channel distortion, and was applied as an inner code. The OFDM signal contains the data in the frequency domain and we applied the trellis coding in both frequency domain and time domain.

The output data, from the interleaver, is encoded by the convolutional encoder shown in Figure 11 which is an Ungerboeck code.

As shown above, the pilot symbol can be used to equalize channel characteristics. We can also use this channel characteristic information in trellis decoding. The pilot symbol, containing channel characteristics, is extracted and forwarded to remodulate trellis reference metric. This remodulation technology can be used to realize both channel equalization and the metric calculation for trellis decoding. Figure 12 shows the bit error rate performance of 64QAM/COFDM using a pilot remodulation metric in the trellis decoding, compared to the trellis decoding, to differential decoding at 12dB of D/U. Using trellis coded 64QAM/OFDM we can improve the bit error rate performance over pilot differential decoding of OFDM. The BER of COFDM is improved by 0.5dB by remodulation of the trellis metric. Although the improvement of C/N is not dramatic in this channel model, we can expect a further difference in performance when we have a null point or very small level in the received signal, which is always a problem for linear equalization. Using these methods we can avoid null point divergence of channel equalization.

By a concatenation of RS codes and trellis coded modulation, we can get a good performance of bit error rate as in shown Figure 12.

## 7. CONCLUSION

We have proposed a video transmission system with pilot aided COFDM. We have described the realization of the symbol timing acquisition and AFC, using pilot symbols. Further more we have shown pilot differential detection of Trellis-Coded Modulation which

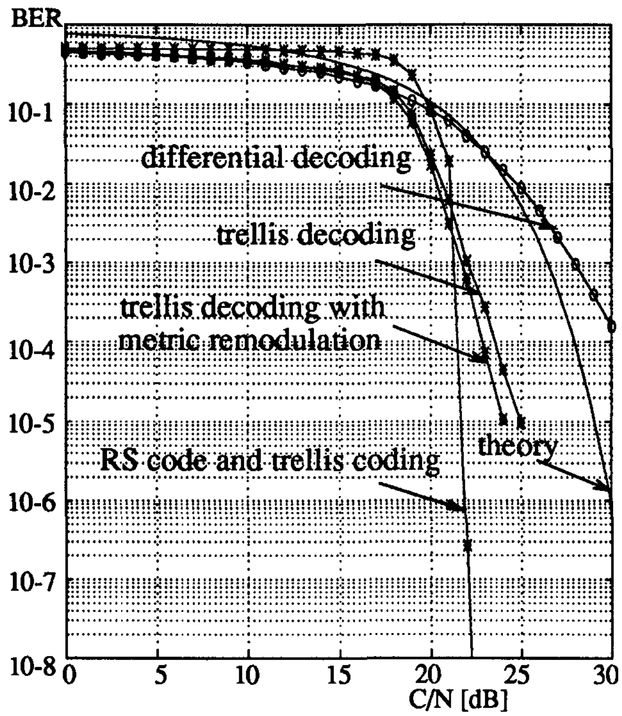


Figure .12 The comparison of bit error rate performance for differential decoding, trellis decoding, trellis decoding with metric remodulation, and the concatenation of RS code and Trellis-Coded Modulation with metric remodulation

improves the bit error rate performance. Concatenating RS codes and Trellis-Coded Modulation we can realize a good BER performance and a very high transmission rate of 22.8Mbps by using 512 QAM carriers. The capacity can be increased to 25Mbps using 560 QAM carriers. We are currently developing the system for laboratory experiment and we plan to compare the performance with the simulation results.

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# EXAMPLE INTERCONNECTION OF ARCHITECTURES, CABLE AND TELECOMMUNICATIONS SYSTEMS

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Bellcore

## INTRODUCTION

Industry reports indicate that Cable and Telecommunications System capabilities are planned to be joined or are being joined. Various companies and consortiums have announced that they plan to offer telecommunications type capabilities over modified cable distribution plant or cable type capabilities over enhanced telecommunications plant. One of the early capabilities which many companies plan to offer over modified cable or telecommunications plant is a wireless communications service referred to as Personal Communications Service (PCS). This paper will concentrate on three typical architectures which could be used to provide PCS. The paper will discuss the interconnection of these typical PCS networks with Local Exchange Carrier (LEC) network. Personal Communications Service (PCS) is a term which has different meanings to different telecommunications service providers. Hence, several views of PCS have arisen as the industry begins to launch initial service offerings. This paper will overview these typical interconnections between three commonly discussed PCS architectures and a LEC. These three PCS architectures are:

1. Personal Communications Network (Diagram 1) - This architecture is being used by some cellular system operators and is being considered by start - up PCS companies who are planning new telecommunications infrastructure for PCS.

2. Cable Systems (Diagram 2) - This architecture is typical of the planned deployment of PCS in some cable systems.

3. Bellcore Proposed Arrangement (Diagram 3) - This architecture has been proposed by Bellcore as a modularized approach to network elements which could be interconnected with a typical Regional Bell Operating Company.

In order to appreciate the commonality of interconnection between PCS and a typical LEC, a review of the commonality of the industry (as a whole) and consumer perspective of PCS is in order.

The FCC (in Docket 90-314, para. 24) defined PCS as a service that will "encompass a wide array of mobile, portable and ancillary communication services and ancillary communication services to individuals and businesses and (will) be integrated with a variety of competing networks.

Early consumer survey responses[1] describe PCS services having some of these attributes:

PERSONAL

*Peace of Mind* - Being able to reach friends and family members at a moment's notice, around the clock.

*Flexibility* - Having the capability to communicate across home, mobile, and office environments using just one phone number.

*Accessibility* - Controlling how much access users want and need.

BUSINESS

*Increased Productivity* - Turning down - time into useful time, around the clock.

*Customer Service* - Handling consumer requests instantly.

*Flexibility* - Never being out of touch.

The overall attributes which a wireless service provider(s) should have the capacity to provide to meet these customer needs include:

Personal number, so that a user can be reached independent of physical location.

Seamless operation, so that a user can enjoy uninterrupted service as they move from one service providers area to another.

Service profile portability, so that a user can move between service providers and retain the same features, such as calling class of service, dialing patterns, screening options and billing arrangements.

Wide area of service, so that a user can move easily on a local or long distance basis.

Screening capability, so that a user can control incoming calls and therefore, their accessibility.

These capabilities are provided to a greater or lesser degree by each of the architectures described. Due to regulatory and legal limitations and business arrangements, the typical PCS network, regardless of architecture, initially will not provide all of the attributes described. Therefore, interconnection with other networks will be necessary to provide services which customers are asking for, based on market information. Moreover, since the existing landline (LEC) network has a large number of subscribers (approximately 145 million access lines), the ability to connect to this large body of users is important to other networks.

Interconnection compatibility information for a LEC to Wireless Services Provider interconnection is described in TR-NPL-0001452. The type of interconnection that is used depends on WSP needs and availability from the LEC, as it is a matter of negotiation between the WSP and LEC. Interfaces which provide the basic interconnection to these three architectures is described in this document. An overview of these example interconnection types is the subject of the remainder of this paper with comments as to how these interconnection types would apply to the three architecture types.

## INTERCONNECTION TYPES AND INTERFACE WITH PCS ARCHITECTURES

### Type 1

The Type 1 interface is at the Point of Interface (POI) of a trunk between a Wireless Services Provider (WSP, which includes PCS services providers) and a LEC End Office (EO) switching system. The WSP establishes connections to the directory numbers served by this LEC EO an other carriers through this interconnection arrangement (Diagram 4).

Incoming calls are handled through the Type 1 interconnection using Multifrequency (MF) trunk signaling protocols. With this Type 1 interconnection, the WSP can establish connections through the LEC network to valid office codes (NXXs) within the LEC local network, LEC Directory Assistance, LEC operator assistance or services provided by Interexchange Carriers (ICs), International Carriers (INCs) and other wireless services providers or local exchange carriers. Outgoing calls from the LEC switched network to the WSP are handled through the Type 1 interconnection using MF trunk signaling to identify the called wireless customer station number without manual or operator assistance.

Type 1 can be applied to the interface for the PCN architecture (Diagram 1) as described above. The Cable architecture can use a Type 1 interconnection from a suitably equipped base station (Diagram 2). Type 1 can be applied, in addition at the Network Interface to the switch (Diagram 3).

The Type 1 Variation interface is based on a National ISDN arrangement. This interface is based on either a Primary Rate Interface (PRI) or Basic Rate Interface (BRI). Other references which will assist the reader in understanding the ISDN interface include SR-NWT-0019373 , SR-NWT-0021204 , TR-NWT-001268 5 , TR-TSY-000268 6 .

Type 1 Variation can be applied to the interface for the Bellcore PCS architecture (Diagram 3) between the Radio Port Control Unit (RPCU) and the Switch. ISDN provides the out of band signaling capability which provides the capacity for the protocol transfer to assist in the network arrangements required for the switch - controller - port interconnection.

Type 2

The Type 2A interface is at the POI of a trunk between a WSP and a LEC tandem switching system. Through this interconnection arrangement, the WSP can establish connections to the LEC EO and to other carriers accessible through the tandem (Diagram 4).

Incoming calls are handled through the Type 2A interconnection using inband MF trunk signaling and trunk address signaling protocols. With the Type 2A interconnection, the WSP can establish connections via the LEC network to valid local network area office codes (NXXs) accessible through the tandem or services provided by ICs, INCs and other WSPs or LECs associated with the local network area.

Outgoing calls from the LEC to the WSP are handled through the Type 2A interconnection using trunk address signaling protocols and MF signaling for identification of the called wireless user's station. Calls are normally routed to the POI based on the NPA and NXX. Shared NXX arrangements with Type 2A are not common and require special translations for routing.

The Type 2A interface can be used, for example:

- As shown in Diagram 1 (or with SS7 in an out of band signaling variation with a Type S - see below)

- With a suitably equipped Base Station in Diagram 2

- At the Network Interface in Diagram 3

The Type 2B interface is at the POI of a trunk between a WSP and LEC EO switching system. The Type 2B interconnection may only provide connections between the WSP and Directory Numbers served by the one EO to which it is interconnected. A Type 2B interconnection may be used in conjunction with the Type 2A interconnection on a high-usage alternate routing basis to serve high-volume traffic between the WSP and the LEC EO (Diagram 4).

Incoming calls are handled through the Type 2B interconnection using trunk address signaling protocols and MF signaling to identify the called station number. With this interconnection, the WSP can establish connections with customers or carriers (e.g. Feature Group A - FGA - IC or a WSP using a Type 1 interconnection) served by DNs in the LEC EO to which it is interconnected. In contrast to the Type 1 interconnection, Type 2B should not be used to route WSP calls to FGB, FGC or FGD ICs or to ICNs.

Outgoing calls from the LEC EO to the WSP are handled through the Type 2B interconnection using trunk address signaling protocols and MF signaling for identification of the called WSP station. Calls routed to the POI based on the NPA and NXX, or 1000s block, if required.

The Type 2B interface can be used at the interconnection point shown in Diagram 1 and at the same PCS architecture points noted for the Type 1 interface for Diagram 2 and 3.

## Type S

The Type S (Signaling) interface is a physical SS7 signaling link connection between a LEC network and a WSP network. The 's' in Type S indicates that signaling information is passed via this interface. The Type S interface is used between a LEC and a WSP to exchange SS7 ISDNUP and SS7 TCAP messages to support the applications to be provided between the WSP and LEC networks. The physical interface specifications for the Type S interface are based on Section 6 of TR-TSV-0009057. The Type S interface may also be use to pass information to other networks. The Type S interface is a physical interconnection. Functions or applications provided by a LEC for a WSP or by a WSP for a LEC would ride over this interface. Specific application implementation will vary among LECs.

Examples of applications which could use the Type S interface include call set up (over a Type 2A or 2B interface, mentioned above). The Type S interface would require a Type 2A or 2B as well to function. See TR-NPL-000145. See Diagram 4. PCS architectures which could use this type of application include Diagram 1, as shown and Diagram 2, where a "Base Station" element in the architecture would have to be equipped for common channel signaling. This application could also be used at the Network Interface in Diagram 3.

Another possible application of the Type S interface is a signaling only interface, such as to transport roaming protocol for authentication and validation functions within the combined networks. A protocol which could be used for this type of function is IS-418. This example is shown in Diagram 1 and 4. A modified IS-41 protocol is used for this type of function in the Bellcore architecture and the Type S could be employed at the Database Interfaces in Diagram 3. Also see References[9,10,11,12,13]

## Summary

Example interconnections have been presented here for typical architectures planned to deliver PCS capabilities. PCS architectures were examined, since these services have a high profile, currently in the industry and are services which many cable operators have discussed as initial or early telecommunications capabilities over their networks.

In conclusion, the three typical PCS architecture arrangements being discussed in the industry today have a number of commonalties for call transport and signal processing.

These commonalties are beneficial for the industry, in that a number of network and architecture combinations can be interconnected, as discussed. This will support an expanded level of services to the customers of all of these carriers.

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# Personal Communications Network (PCN) Concept

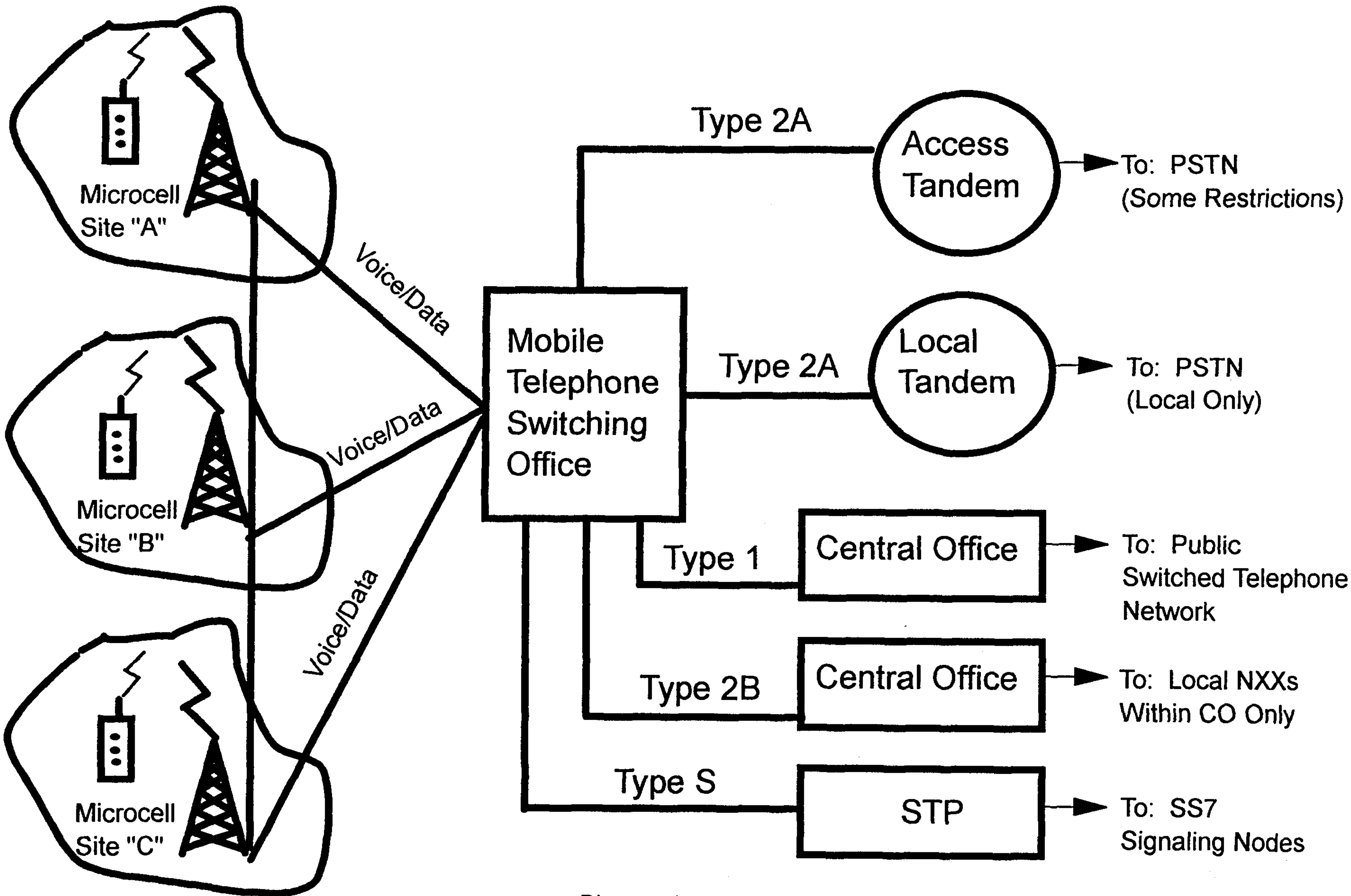
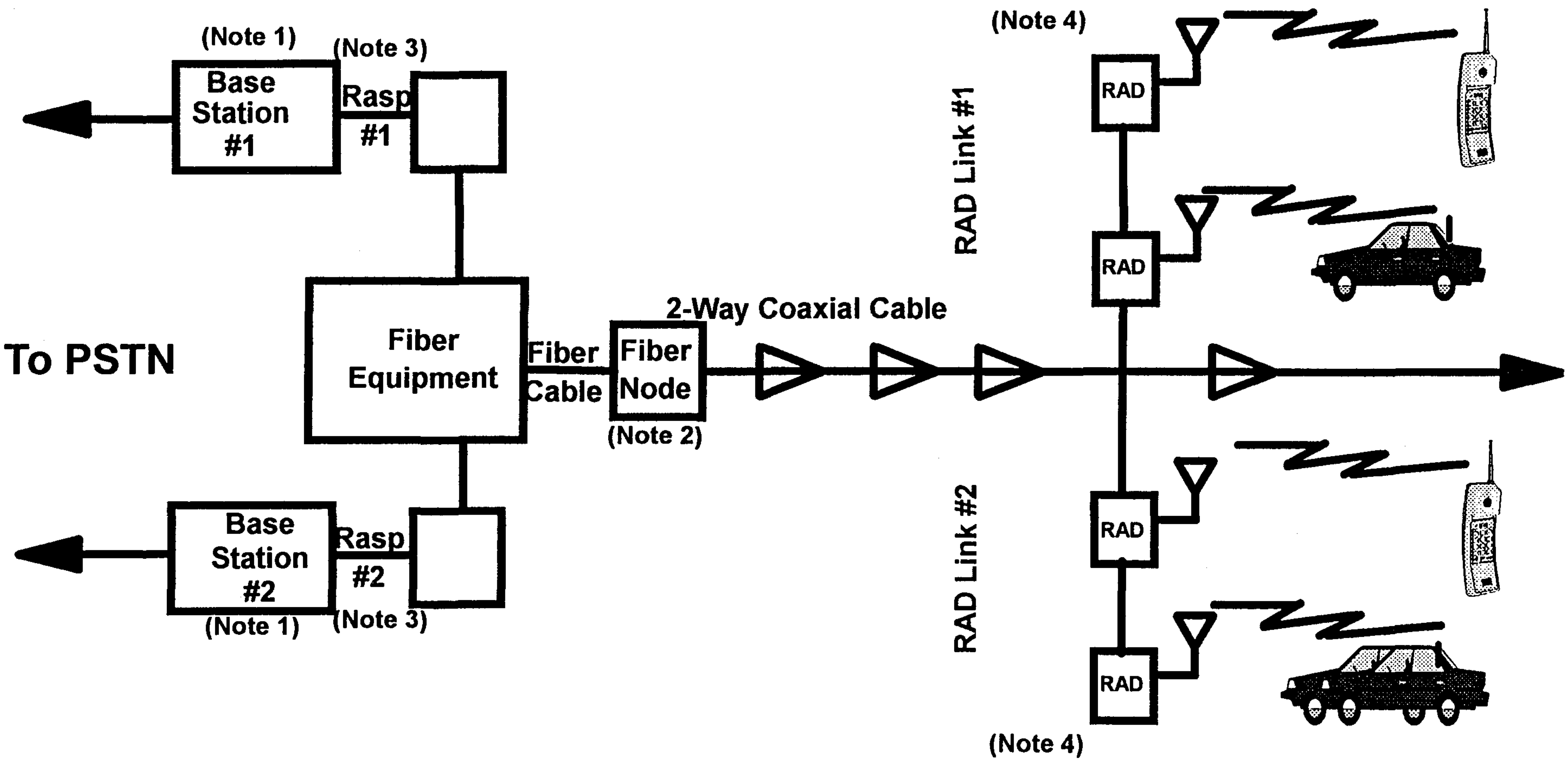



Diagram 1

# Generic Cable Architecture for PCS

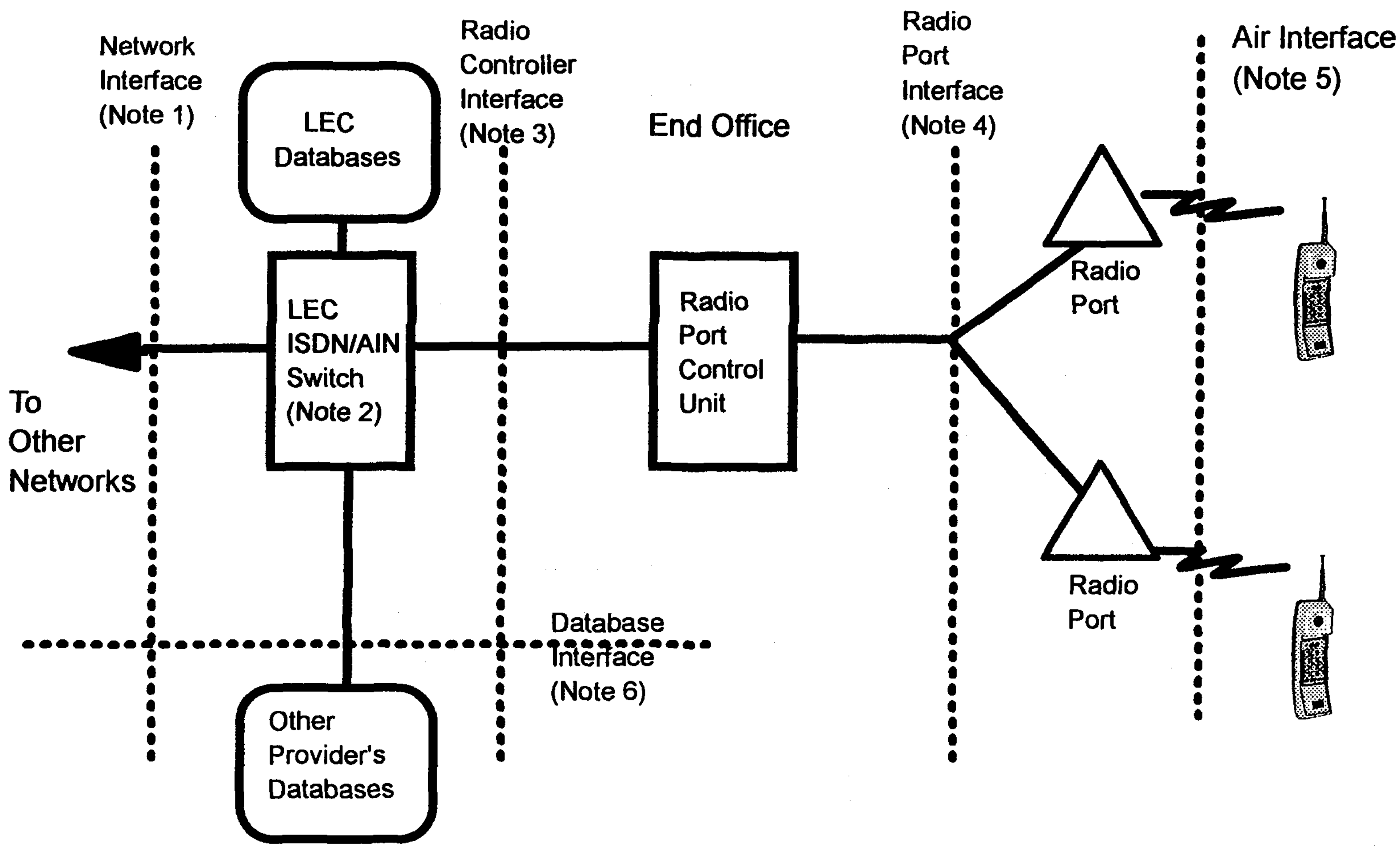



Notes:

| | |
|---|---|
| (1) Multiple Base Station: | Provides channel allocation, power control, handoff for radio channels and interface to PSTN. |
| (2) Fiber Node: | Contains equipment to convert optical signals to electrical. |
| (3) Remote Antenna Signal Processor (RASP): | Controls Remote Antenna Drivers (RADs) via signals from Base Station. |
| (4) Remote Antenna Drivers (RADs): | Remote antennas providing coverage within radius of approximately 500-1,500 feet. Each series of RADs emanating from the RASP simultaneously transmit identical frequencies. Handoff between RAD Link #1 and RAD Link #2 is possible. Handoffs are controlled by Base Station via signals to RASP. |

Diagram 2

# Simplified Bellcore Proposed PCS Architecture




NOTES:

(1) Includes any switched interconnection to PSTN, e.g. Type 1, Type 2, DID and SS7
(2) May be several switches. Must have ISDN capabilities. AIN and SS7 features need further development.
(3) Standard not yet formalized. Initial Bellcore proposal uses ISDN-BRI connection.
(4) Standard not yet formalized. Link may be leased analog, digital, or fiber. Initial Bellcore proposal is TI, DSL, or DLC links.
(5) Standard not yet formalized. Bellcore proposes using TDM/TDMA per TA-NWT-001313.
(6) Database owned by another provider. Signaling link is 56kbps data line.

Diagram 3

# Interconnection Examples Overview

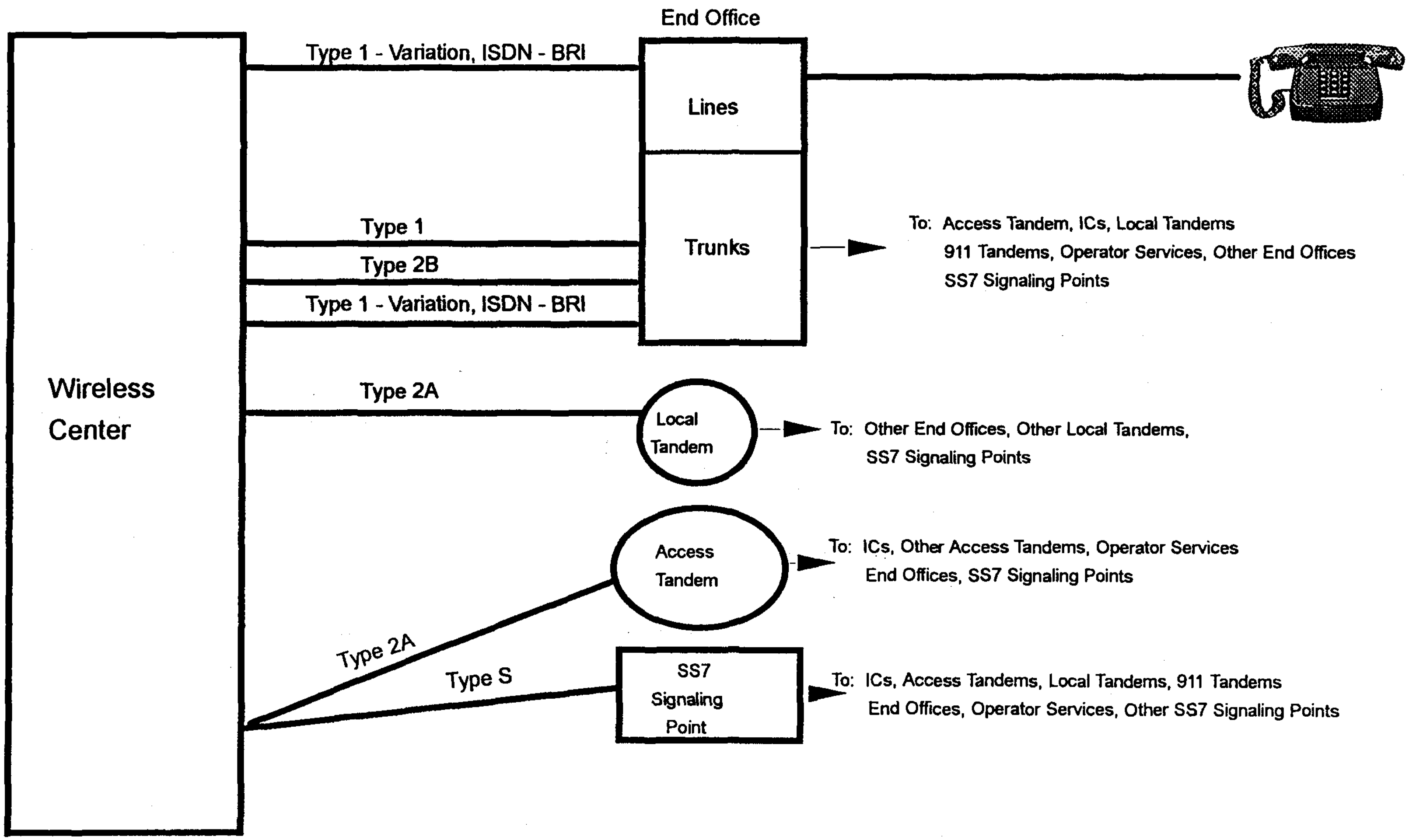



Diagram 4

# Fault Tolerance in the Orlando Full Service Network

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## Abstract

This paper discusses the notion of fault tolerance in an interactive multi-media delivery system. The various components are reviewed together with strategies for hardware redundancy and the software mechanisms necessary to support the use of hardware redundancy. Traditional telecommunications approaches to fault tolerance are referenced and the cost and complexity of these schemes is shown to be unrealistic for the kinds of full service networks envisioned. The Orlando Full Service Network is described and its expected and actual failure modes are discussed. Fault tolerance mechanisms that are designed to use spare capacity in the delivery system are described and the effectiveness and simplicity of alternative schemes are discussed. The paper proposes a set of network design rules that can be used to build fault-tolerant entertainment-delivery networks without increasing cost dramatically.

## Introduction

The Orlando Full Service Network is designed to provide interactive television services (including movies-on-demand, home-shopping and video games) to a community of 4,000 subscribers. In a system of this size, it is important to understand the possible failure modes and their impact on system availability. To meet service availability metrics, it is important that certain faults can be tolerated by the system to eliminate or reduce down-time due to failures. Thus we have a requirement for a Fault Tolerant system design.

### System and Server Availability Requirements

When designing any system certain requirements for system and service availability can be defined. There are a number of ways of doing this but two of the most common metrics are MTBF and MTTR.

Mean Time Between Failure (MTBF) is a way of measuring how often (on average) a component or the entire system can fail. Typically, in a product development environment, the MTBF of each component is modeled and the expected MTBF of the entire system can be predicted statistically.

Mean Time To Repair (MTTR) is another important metric, because after a failure has occurred, the key to returning the system to service is the time (on average) it takes to repair the system.

Using MTBF and MTTR metrics allows the designer to predict the expected availability of the system and of the service that the system provides. If certain system failure modes only affect part of the system, then the service availability will be different from the system availability. For example, in the FSN a failure of a single server may cause a loss of service to a fraction of the subscribers, but not impact the other subscribers at all.

### Resource Management Requirements

In a complex network like the Orlando FSN, there are many failures that can and will occur. These may be as simple as a loose connector, or as complex as a software bug in a Media Server. In the event of such a failure, a well designed system will respond in two ways:

- The system will attempt to use other resources to replace the failed resource. For example, if a DS3 link fails, the traffic for that link may be re-routed over other DS3 links (assuming there is sufficient spare capacity).

- The system will generate an alarm to notify a technician of the failure. The failed component must be isolated and replaced as quickly as possible to restore the system to full operation. In a complex system this fault isolation step can be very complex to do manually. A network management system can reduce the time to isolate a fault by using a rule-based diagnostic tool. A set of rules is developed over time by the system designers and technicians. The diagnostic tool uses these rules to infer which component has failed and produces a report for the technician on duty.

## System Overview

A system overview diagram is presented below. The key elements that are monitored are the Media Servers, the ATM Switch, the ATM Multiplexers (MUX) and the Home Communications Terminals (HCT).

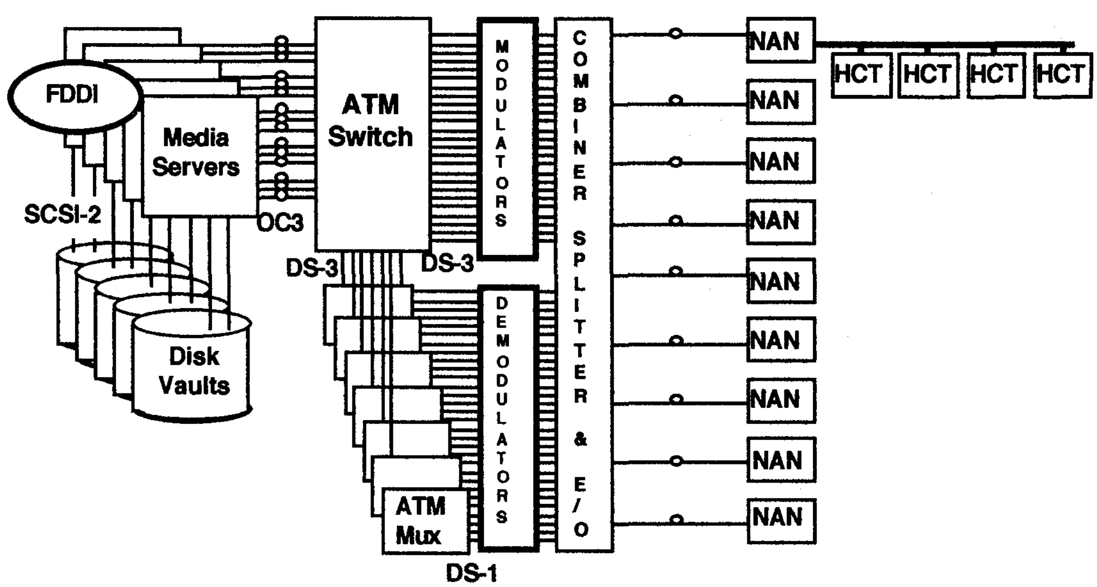


**Figure 1. System Overview**

The major system components are shown in Figure 1. The impact on system availability of each component is described below.

- The ATM Switch is a single device, which provides all connectivity from the Media Servers to the HCTs. If the ATM switch were to fail completely, interactive services would be interrupted to **all** FSN subscribers. (However, analog cable services would still be available).

- There are 7 ATM Multiplexers, which operate independently of each other. Therefore the complete failure of a multiplexer would interrupt interactive services to approximately one seventh of the FSN subscribers. (However, analog cable services would still be available).

- There are 152 QAM modulators. Each serves up to 7 active, interactive subscribers.

- There are 16 QPSK modulators. Each serves an entire neighborhood (providing control and timing information to all HCTs in that neighborhood).

- There are 240 QPSK demodulators. 160 modulators are dedicated to reverse signaling from interactive subscribers and each is shared between 25 subscribers. (The remaining 80 modulators are available to provide services requiring higher bandwidth return, such as video telephony or data services).

All above equipment is located in the Network Operations Center (NOC) and is powered by a conditioned, un-interruptable power supply. Reserve power is supplied by batteries and a backup diesel generator. The headend equipment is housed in approximately 1000 square feet of computer room which is air-conditioned and equipped with a raised floor. Fire protection is provided by a Halon gas extinguishing system.

There are a total of approximately 500 physical interfaces between the various components in the NOC, and thousands of optical and electrical connectors and cables. The ability to quickly isolate and repair any problems as they occur is very important in this complex a system.

- There are 16 neighborhoods (of approximately 250 subscribers each). Each neighborhood is served by a laser transmitter and receiver at the headend and Neighborhood Area Node (NAN) in the field. The NAN provides optical to electrical conversion at the point where the fiber ends and the coax network begins. The failure of a headend transceiver, the fiber or the NAN would interrupt interactive and analog services to all subscribers in a neighborhood. The NAN is powered by conventional, battery-backed power-supplies via the coaxial plant.

- There are 4000 Home Communications Terminals, each located in a subscriber residence. Obviously, the failure of an HCT impacts only a single subscriber, however it is more difficult to service because of its location. The HCT repair strategy is unit-replacement. It is very important that failures in the system that manifest themselves as an HCT not working do not result in a wasted visit to the subscriber.

This is a key difference between a traditional cable system which, being completely broadcast in nature, does not usually experience these kinds of failure. In the Full Service Network, which is much more transactional in nature, it is often not obvious whether a failure is in the HCT or some component in the NOC.

**System Failure Modes**

The preceding discussion describes some of the ways in which the various components that make up the system can fail. It is very important to understand these failure modes because the failure of only part of the system is not necessarily catastrophic as long as:

- **The failure does not affect the rest of the system.** For example, the failure of an HCT would seem to be a trivial case, but a failure mode in which the HCT's reverse transmitter starts to send continuously would impact the other 24 HCTs sharing that reverse QPSK channel. (In this example, the solution is to implement a reset command to allow the system to disable the offending HCT and localize the failure to that subscriber).

- **The failed component can be replaced without impact to the rest of the system.** This may sound obvious, but it is remarkable how certain failure modes seem to appear which require a complete system outage to clear them. These failure modes have a major impact on system availability. It is important to recognize these failure modes in the design phase so that they can be 'designed out' of the system.

For each component that can fail, a decision has to be made: **Is it worthwhile to design-in fault tolerance?** The design philosophy for each major component is discussed below.

### Media Server

The FSN uses eight Silicon Graphics Challenge XL servers. The server hardware is not fault-tolerant. (The cost of providing hardware fault tolerance would more than double the cost of the media servers). However, the media servers form a fault-tolerant distributed computer system. At a software level, redundancy is provided in each server to spare the others. The design of such a system is complex, however not all functions need to be duplicated. Special design attention to redundancy was given to following critical components

Connection Manager

The connection manager is an example of a critical function that should not fail if a server fails. The connection manager allocates an ATM Virtual Channel in response to an application request. To make the connection manager reliable even when a server fails, it is split into Neighborhood Connection Managers (NCM). Each NCM is duplicated in software such that the backup NCM runs on a different server than the active NCM.

If the active NCM fails (because the server hardware fails), the backup NCM can take-over without loss of service. When the server recovers (usually after a re-start), the active NCM is restarted and resumes its normal operation. To allow the changeover to be seamless, the active and backup NCMs checkpoint the current connection information to a shared file.

Software Processes

Any software process may fail due to a hardware or software exception. The general strategy is to restart any process that fails to recover from exceptions. Typically each server has hundreds of software processes running at any moment in time, and the failure of a single process should not affect the integrity of the others. This is achieved by using Virtual Memory techniques that provide each process with its own unique address space. Even if the process attempts to write to random memory addresses, the memory management hardware will protect the other processes from being affected.

Disk Vaults

Each server is connected to a large number of disk drives which are housed in disk vaults. The drives are connected using a SCSI-2 interface controller for each 16 disk drives. This approach reduces the impact of the failure of a single disk drive or SCSI-2 controller.

A number of more sophisticated approaches exist to provide for fault tolerance. These are generally known as RAID - Redundant Array of Inexpensive Disks. There are actually five main flavors of RAID from RAID-1 through RAID-5. Some approaches are reliable but store two copies of the data (mirroring), and so require twice the disk storage. Other approaches achieve reliability at lower cost by storing additional parity information that can be used to reconstruct data lost due to a disk failure.

### ATM Switch

The FSN uses a single AT&T GCNS-2000 ATM Switch. As previously noted, the ATM switch is a single point of failure for the entire system. This dictated the selection of a highly reliable switch[1]. Fortunately this is a common requirement in the telecommunications industry and the development of extremely reliable switching platforms has evolved to meet this requirement.

AT&T are experienced in the design of highly reliable switches. (The Bellcore recommendation for ATM switches is a maximum of 2 hours downtime in 40 years! ). The AT&T GCSN-2000 design philosophy eliminates all single points of failure. This is achieved by duplicating all critical components. Each active

[1]. Note that the chosen solution is not the only one available. An alternative is to implement a reliable network based on multiple simplex switches instead of a single duplex switch. Network redundancy (as this is called) has the advantage of using less expensive, simplex switches and of enabling the use of '1 for n' sparing. For example, 10 simplex switches could be spared by an eleventh switch. A further extension is to build a system with 10 switches and accept the reduced impact of a single switch failure.

component is monitored and in the event of a failure, the backup component takes over the tasks of the failed component without any interruption.

Another important requirement is that it must be possible to repair the switch while it continues to operate normally. (This is analogous to replacing a tire on a car while driving it down a freeway at 55 miles per hour). In order to allow this, it must be possible to 'hot-swap' cards while the system is powered. This requires special power-up circuitry and connectors on the cards and card-cages.

In the ATM switch the following components are duplicated:

Control Processor

The control processor is responsible for monitoring the status of the switch and allowing the provisioning of ATM connections. The entire control processor is duplicated and each processor monitors the other. Each is capable of running the entire switch if the other fails.

Shelf Controllers

The Shelf Controller is responsible for monitoring the status of a shelf and allowing the provisioning of ATM connections. The shelf controller is duplicated and each shelf controller monitors the other. The links to the control processor are also duplicated.

Fabric Interface

The interface from the shelf to the fabric is duplicated. However, each fabric interface is only connected to one of the two switching fabrics. This means that the system can not tolerate two fabric interfaces on two different shelves. This is a double failure and is extremely unlikely in a given time window.

ATM Switch Fabric

The ATM Switch fabric is duplicated and each fabric is capable of supporting the full traffic capacity of the switch. In the event of a failure of either fabric, the other fabric is able to take over with the loss of only a small amount of traffic. In practice, we found the traffic loss would not normally be noticeable to a subscriber viewing a movie.

Clock Distribution Cards

The Clock Distribution Cards are duplicated and monitored by the shelf controllers. If the active card fails, the standby card takes over.

Line Interface Cards

In addition, the Line Interface Cards can be duplicated, but in our application this was not required because the loss of a single line card does not affect a large number of subscribers. (Approximately 100 subscribers would be impacted (worst-case) by the failure of an OC3c line card).

The physical layout and location of the components is illustrated in the diagram below. Note that the physical dimensions of the switch are significantly larger than for an equivalent simplex switch.

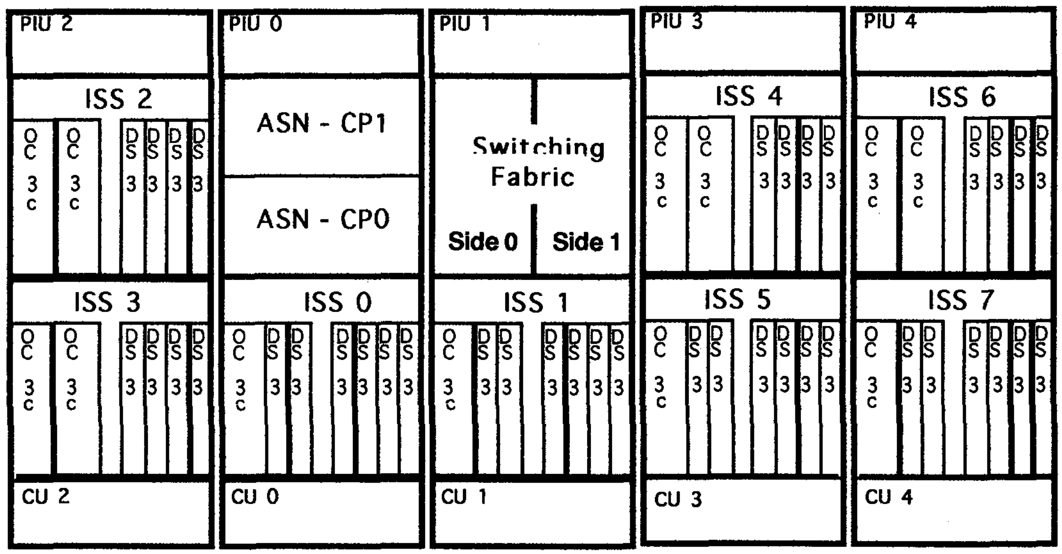


**Figure 2. ATM Switch Physical Layout**

### ATM Multiplexer

The FSN uses seven Hitachi AMS-5000 ATM multiplexers. The multiplexer is responsible for the forward control and reverse application channels to 2 or 3 neighborhoods (400-600 subscribers). As such, it must be reliable and should not be subject to a single point of failure. The multiplexer has the following redundant components:

Control Processor

The control processor consists of 2 cards and can be duplicated. In our application this was not considered necessary because the multiplexer will continue to handle traffic even if the processor fails. (Only the provisioning and monitoring functions are affected).

Backplane Interconnect

The traffic is switched from one interface card to another by being routed over a shared-bus backplan. Each interface card acts as a repeater in the bus. To prevent the failure (or removal) of a card from interrupting the traffic, the bus can dynamically re-configure around a failure.

Interface Cards

The interface cards are not duplicated in this system. The failure of a single DS1 transmitter card could affect interactive services on up to 3 neighborhoods. New options for making this card redundant are being considered by the Orlando personnel.

The failure of a single DS1 receiver card impacts approximately 100 subscribers and is therefore not considered critical.

### Modulators and Demodulators

All of the RF modulators and demodulator are supplied by Scientific Atlanta.

Each QAM modulator serves up to 7 active subscribers at any point in time. If a modulator fails, only 7 subscribers would be affected. An exception to this is the modulator used to broadcast the kernel software to the HCT. This is a single point of failure in the system but it is not duplicated because it only affects the ability of HCTs to boot.

Each QPSK modulator serves an entire neighborhood (about 200-300 subscribers). The QPSK modulator is not duplicated.

Each QPSK demodulator is shared between about 25 subscribers. The QPSK demodulator is not duplicated.

### The Laser Transmitters

The laser transmitters and receivers are supplied by Scientific Atlanta.

There is a single Laser Transmitter for each neighborhood. The optical signal may be split 2 or 3 ways to the physical nodes in the plant. The Laser Transmitter is not redundant. If it should fail it would typically affect about 200 to 300 subscribers.

### The Neighborhood

The neighborhood is not a fault-tolerant unit. This follows the tradition cable system view for upgraded plant. The components from the laser transmitter in the head-end to the NAN are all of low complexity, and the transmit path is analog. To duplicate the equipment and fiber, and to provide protect-switching (as SONET does) would increase the complexity and cost to an unacceptable degree.

### The Home Communications Terminal (HCT)

The HCT has no requirement for fault-tolerance as it provides service only to a single subscriber. However, it must be reliable so that it does not significantly reduce the service availability to the subscriber. It must also not affect any other subscribers if it fails. This requirement is partly satisfied by the coaxial network itself which includes resistive power taps that prevent event a short-circuit at the drop from affecting other subscribers. However, if the HCT starts to transmit continuously it would impact the other HCTs sharing the reverse QPSK channel. In this case, all HCTs sharing the reverse channel will be effectively jammed, and the regular heart-beat exchange between the Server and the HCT will fail. This failure will be reported to the Network Management System.

### Network Management

The inclusion of a sophisticated Network Management System (NMS) in the FSN does not affect the MTBF of any service-related component. However the NMS helps to reduce the MTTR when a failure occurs. The NMS does this in two ways:

- By monitoring the system components, an alarm indication can be generated as soon as a failure occurs.
- By building a set of rules into the NMS, the NMS can greatly assist in Fault Isolation. This reduces MTTR by helping focus repair activity on the correct component. In a complex system it can be extremely difficult to locate the failed component without this kind of assistance.

### Power Supply and Air Conditioning

The physical and electrical environment in which the system operates is often overlooked. The Orlando Network Operations Center was designed to meet Telco standards which include reliable systems for:

- Un-interruptable Power Supplies - this includes battery power and a backup generator. In a normal telephony operation, all systems are required to run from a - 48 volt supply. In the FSN, only the ATM equipment meets this requirement and so invertors are required to support the Media Servers, Modulators and Demodulators and AM Laser equipment.
- Air Conditioning - this must also be supplied with backup power or temperature rise would cause equipment to automatically power down after a number of hours (depending on ambient temperature).

## Conclusions

The system described is now in commercial service in Orlando, Florida. The metrics of service availability, MTBF and MTTR are being tracked for the various components. This information will be extremely valuable in the design of future networks. Designing fault tolerance into a system is only justified when failure rates and loss of service impacts are greater than a certain thresh-hold. This thresh-hold also changes with the set of applications and the subscriber's expectations, which are both in a state of flux.

Future applications (such as Telephony and Data Communications), have different service availability requirements from interactive television. The experience gained in Orlando will be especially valuable in constructing future networks to deliver a wide range of communications and entertainment services.

# FIBER BY DESIGN

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*Abstract*

*The evolution of CATV systems employing fiber optics has been both rapid and dramatic. Fiber optic CATV use has made the transition from following and adapting technology developed for telephony to driving fiber optic development. Couple this with the continual and fast changes in fiber optic technology and it is not surprising to have conflicting information when designing a CATV system. This paper presents a review of the major fiber optic technologies and gives some guidelines that can be used in determining the optimal design.*

## INTRODUCTION

The rapid development of CATV fiber optics has led to some confusion and contradictory information in system implementation. This paper reviews the technology and offers some guidelines in system design. To review and compare technologies for fiber optic CATV it is apparent that system aspects that are discriminators be brought out. Rather than listing the advantages and disadvantages of competing approaches this paper will present the desired system goals and see how alternative approaches rate within that category. However, these options easily fall into two classes. The first is the forward (headend to node) system including the Optical Transition Node (OTN) method of remotting headends and/or reducing fiber counts. The other is the return (node to headend) system, and this separation will be used for the following discussion.

## FORWARD SYSTEM

Today the CATV forward system mainly carries video channels. However the system is expected to carry video, telephony, personal communications and data. The main technologies, in practical use, for the forward system are 1310nm DFB transmitters, 1310nm Externally modulated transmitters, and 1550nm Externally modulated transmitters. Each of these offers unique advantages and could be considered for a CATV forward system. This paper assumes that receivers do not make a critical difference in the system tradeoffs. This assumption is justified, in that the receiver semiconductor, InGaAsP, is capable of receiving both the 1310nm and 1550nm wavelength. Also, once most system parameters are determined the receiver choice would likely be the same for all the potential transmitters. In the next few paragraphs system considerations are presented and the impact by differing technology choices is considered. One issue at a time will be considered without regard to others. These issues will then be tied together at the end of this section.

### Cost

No system design can be considered without the overriding concern being cost. It is imperative that the completed system be profitable to operate. The system cost is not just the price of equipment and installation fees but involves maintenance as well as reliability. The last two could conceivably be the cost drivers over the expected system life.

Certainly, in cost, it can be seen that the 1310nm DFB transmitters have the advantage. The 1310 DFB transmitter is the most commonly used transmitter in the CATV plant. This has translated into volume related discounts that are passed on to the customer. Also it has a proven reliability record that comes with the maturity of the device and product offering.

The 1310nm External modulator is the next choice. This product offering is relatively young and the volumes are not on the same order of magnitude as the 1310 DFB transmitter. It does, however, take advantage of proven techniques and components, to a large extent, indicating the potential for highly reliable operation.

In cost the last choice is the 1550 External modulator. This offering is in its infancy and while interest is growing it is still a low volume product. From the point of view of maturity there are major components that are relatively new and this is a potential risk.

Link Length and Performance

The transmission length in fiber optics is specified in dBs of loss. This is made up of two components. The first is passive loss (couplers, splitters, wavelength division multiplexers) that will be considered equal for both wavelengths. The second is fiber loss and is specified as dB/Km at a particular wavelength. For wavelengths near 1310 this is approximately 0.35 dB/Km while for 1550 this is 0.22 dB/Km. Based on fiber loss alone there is a significant advantage to 1550 operation. However, since real systems always have a component of passive loss the actual length of fiber is always less than would be predicted by fiber loss alone. The real advantage in link length for 1550 lies in the opportunity to repeater the signal with an inline optical amplifier. The optical amplifier also offers the advantage of transparency. Essentially this allows system upgrades at the transmission and reception points without needing to change intermediate repeaters. Currently, the 1550 fiber amplifier has no competition in the 1310nm operating window. While 1550nm External modulators have a distinct length advantage (when optical amplifiers are considered), the second choice is the 1310 external modulator. Since the source can be a solid state laser, high power operation is limited only by the modulator physics. It must be noted that this operational limit has yet to be met in a practical application. The third choice in link length is the directly modulated 1310nm DFB laser. However, the range of this choice is easily sufficient for 90% of applications and is also being improved on a yearly basis.

In link length, the issue of Stimulated Brillion Scattering (SBS) often arises. This is a nonlinear interaction of acoustic waves in fiber. The limit for a purely coherent source is on the order of 10mW or 10dBm at both wavelengths (the limit is wavelength dependent and is lower at 1550nm). However the SBS threshold is affected by the dynamic linewidth of the source. For directly modulated DFB lasers the limit is an order of magnitude larger than 10dBm and is therefore not an issue in CATV systems. For externally modulated transmitters the SBS limit can be a problem, however by "dithering" the source (via phase modulation) the limit can be raised to the needed level eliminating any concern.

You cannot separate reach from distortion performance as an issue in system design. If you were to separate them the comparisons of differing technologies would be meaningless. So in the preceding paragraphs a standard 77 NTSC channel lineup with 200 MHz of digital loading has been assumed. In this a C/N of 51, CSO of 62, and CTB of 65 for the video has been used. However this is not the whole story. While both the direct modulation and external modulation techniques use complex linearization schemes the external modulator is theoretically easier to linearize and therefore some specifications have shown an improved CSO to 65 dBc. This is important for repeatered applications and an example of this is remotting headends via the OTN approach. For systems (with or without transition nodes) that are approaching the "passive coax" model, by having less than 6 rf amplifiers in cascade, this is a non issue. Indeed these systems are normally C/N and/or CTB limited and CSOs of 58-62 dBc are acceptable.

## Narrowcasting

For this paper narrowcasting will be defined as the need to provide a single transmitter per receiver allowing different channel lineups and data to smaller groups of customers while increasing system reliability. It has its basis in the desire to run telephony over cable. Traditionally a transmitter output is split providing the same signal to 4 optical nodes. Therefore narrowcasting has a direct impact on cost because there are now approximately four times the transmitters per system. For this definition the 1310 DFB transmitter is the desired choice, it is the most cost effective solution with a proven reliability track record. The external modulator choices of both wavelengths have their advantage in serving multiple receivers, and this is particularly true for the 1310 External modulator.

## Choice

As can be seen in the preceding paragraphs the best choice depends on the specific system needs. The three extreme examples that follow help define the boundaries of this decision.

The first system is the typical CATV system. This system is in a suburban area with customers within a 20 km fiber radius of the headend. It is important to be able to provide standard CATV service today with the option to upgrade to Narrowcasting to compete with the local telephone company. In the preceding paragraphs we see that the needs today are low cost, short fiber length and CATV reliability, while the needs tomorrow are low cost/node, signal diversity, short lengths and telephone reliability. The difference between the "reliability" considerations results from telephony being considered a lifeline service. When we compare these needs to the above discriminators we see that the 1310 DFB transmitter is the obvious choice.

The second system serves a highly dense urban area with extremely short fiber lengths but many splits from a single headend. This system is looking for as few components as possible, with low cost/node, but due to density it will be sending the same signal lineup to many nodes. This is a bit trickier than the previous system in that it seems the DFB transmitter might be the preferred option. However the intent is to show that even though the fiber lengths are very short the loss due to extensive splitting makes the overall loss budget large. With this in mind the preferred choice is the 1310 External modulator. Even though this unit cost more than a DFB, it is ideally suited for this multiple splitting application.

The last system is in a very rural area where fiber lengths are long and headends are remoted from the customers for ease of maintenance. The system has been determined to be profitable with any technology, but the overriding concern is reach. In this scenario the best alternative is the 1550 external modulator coupled with fiber amplifiers as needed.

In these three polarized systems the choice is obvious. In reality any moderately large system may include all of the above scenarios. Therefore a system may need more than one technology or may make tradeoffs in technology use to accomplish its goals. All the technologies have their place for the foreseeable future and the choice on which is best is system dependent.

### Future Intangibles

The rapid advancement of fiber optic technology is making our industry change even faster than the personal computer industry! Therefore advances that could change the product and technology mix are inevitable. The most interesting change that could happen is the development of a 1310 optical amplifier. The transparency at 1310nm coupled with the overwhelming use of this wavelength in CATV fiber optics would easily shift the decision lines.

## RETURN SYSTEM

The return fiber optic system is identical to the forward system in that cost, reliability, maintainability, link length and performance, and narrowcasting are the key system concerns. Unlike the forward system, the return system topology is not well established, and this leads to additional confusion. This portion of the paper takes the same style as the preceding section to help in the design considerations of return systems.

The CATV return system carries mainly FSK converter data as well as the odd video channel today. It is expected to grow completing the bi-directional path for video data and telephony. Each return band is capable of transmitting 6 video channels with bandwidth in the 5-42 MHz region or many data carriers, dependent on modulation type and bit rate. This capability is augmented at the fiber node by taking advantage of the unlimited fiber bandwidth. Typically four of these return bands will be combined on the return fiber link. The infancy of this application has brought a multitude of potential solutions from 1310 and 1550 nm FP lasers, to 1310 and 1550 nm DFB lasers. Each brings its unique advantages to the fold, but once again we shall look at the system needs and compare the choices.

### Cost

Certainly the application of the return system dictates the cost of the return transmitter and whether the system can generate revenue. Therefore the following information must be held until further system needs are considered. The 1310 FP laser is the lowest cost device, followed by the 1550 FP laser, the 1310 DFB, and the 1550 DFB. However these costs differences may not be as extreme as first thought. Since the return system need not be as complicated as the forward (80 Video vs. 2-12 Video) and the number of return transmitters needed today is roughly 4 times as many as are needed in the forward system (due to outbound splitting) the variations in cost can be lower that expected. In terms of maturity the order presented above needs modification in that the 1310 DFB and the 1550 FP trade places. This opinion is due to the fact that there are few applications for 1550 FP lasers outside instrumentation.

Link Length and Performance

This is generally where confusion in the return system arises. A typical system returns fiber to the headend at a much shorter optical length than is transmitted from the headend. The reason for this is the forward transmitter is normally split to multiple receivers. This splitting loss is not actual distance and is not encountered in the return path which will have one transmitter per node. In any case, the loss of fiber at 1550 nm is still markedly lower than at 1310 and this is always an advantage.

For lower level data applications (like FSK converter data) all the technologies mentioned can readily be used; however, the FP lasers are the devices of choice due to cost. Using a 1550 nm FP results in a high noise environment due to the interaction of the laser mode partitioning and the fiber dispersion. Data application may still be valid though, and this device cannot be ruled out. It would seem most appropriate to use this device when length is at a premium.

For low numbers of video (2 channels typically) the 1550 FP is ruled out due to the noise, as mentioned above, and the others can still be considered. Here length again plays an important role in determining the choice. The FP is good for short lengths while the 1310 DFB can be used for longer lengths and the 1550 DFB for extremely long lengths.

For large numbers of video channels the FP lasers is completely ruled out and only the DFBs remain. Their use is predicated upon length and channel line up. The 1550 DFB is possible here and more information will be gained by a paper presented in this session, "Return Path Lasers for High Capacity Hybrid Fiber Coax Networks."

For hybrid applications of multiple video and data channels, as would be used in a telephony application, the choice would seem to favor the DFBs again with emphasis on the 1310 DFB.

It should be mentioned that the use of a DFB is not without certain costs. They include a large power consumption for the thermal electric cooler and the high packaging costs. The application needs to justify this cost, and surely the revenue from telephony could do so.

Narrowcasting

Narrowcasting does not directly effect the choice for the return system as it does in the forward system. The reasoning here is the return system is not split and therefore does not see an impact. Hidden in this, however, is that narrowcasting drives the DFB volume higher and this can only foster lower prices for all grades of DFB. Therefore narrowcasting can help make the DFB return transmitter a profitable solution.

### Choice

Again, the choice is application dependent. Since applications are too diverse in the return system only two extreme examples will be given.

In the first example the system operator wishes to return FSK converter data over short link lengths. In this application a 1310 FP would be the best possible choice. The cost is the lowest with high reliability. Distances of 7-10 dB are easily handled.

In the second example. The system operator is in head to head competition for telephony. The franchise agreement also requires local video origination. The link lengths again are short. In this application the 1310 DFB is the device of choice. With good linearity and low transmitter noise it will be able to handle the job admirably. The higher cost is expected to be offset by telephony revenues.

### Future Intangibles

There are two main areas that can change these concepts. The first is the development of a DFB without the need for an expensive Thermal Electric Cooler. Work in this area progresses, and this could lower the cost of the DFB laser. In addition the main reason that FP lasers are noisy is mode partitioning. Simply stated the FP lasers have many optical carriers all competing in a random fashion for photons. If a FP laser could be developed with a single mode, then this mode partitioning noise would be eliminated. This is being researched, for other applications, as a surface emitting laser. Of the two the wide temperature range DFB is closer to being productized. Fiber optics is changing rapidly, monitoring these changes will allow you to make application improvements as they become available.

## CONCLUSION

This paper has looked at the system design of a fiber optic CATV application and compared various approaches based on system needs. The conclusion that the correct choice is system dependent has been shown. Indeed, a complete system most likely benefits from a combination of all technology choices. The other major conclusion is that fiber optics is a rapidly changing field and these comparisons are only valid as of the writing of this document (March 1995). The continual changes need to be monitored to keep confusion at a minimum and benefit at a maximum.

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# Graphical User Interfaces: The Success (or Failure) to Navigating the InfoBahn

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General Instrument

***Abstract***

*The graphical user interfaces being designed for consumer broadband television devices will play a crucial role in determining the success – or failure – of current and future services. These initial user interfaces will train viewers to navigate the wealth of programming and services available through new technology. An effective graphical user interface will encourage viewers to use the services currently being offered, and to be receptive toward future services. If the user interface makes it difficult to use initial services, the viewer may not be receptive toward new services.*

*This paper will explore these issues..*

## Couch Potato Mode -- Or That's Entertainment

As people spend more time managing work and home responsibilities, they have less to spend in front of the television set. When viewers do turn on the television, they expect to be entertained. They are in the relaxation, -- a.k.a. "couch potato" -- mode. Television is meant to be easy, mindless. Despite hundreds of choices, graphic user interfaces should make locating something to watch effortless. The last thing the viewer expects to see is a DOS prompt on the television screen.

## Choices, Choices, Choices: It's Not Just Television Anymore

Hundreds of channels of program offerings. The power to watch what you want at your convenience. Digital stereo audio and wide screen televisions for home entertainment theaters. Secondary audio programming for multi-lingual households. Downloaded video games, CD-quality commercial-free music programming and home shopping all available at the press of a remote control button. The good news is that convergence of computer and broadband television technologies provides viewers greater variety in television programming *and* services. Viewers have more control over what they watch and how it is delivered.

The not-so-good news is that the video tuning device that enables these choices is more sophisticated. Channel surfing is still possible, but it no longer allows the viewer to maximize his or her viewing pleasure potential. The viewer must interact more frequently with the device, not only to find something to watch or select a service, but to modify the manner in which this program or service is delivered. If the household is Spanish-speaking, the device would display all graphical user interface text in Spanish and to tune all Spanish language audio where available. If the viewer is the only Spanish-speaking member of a household, he would like to enable Spanish language features during his viewing sessions only. Technology being developed enables these capabilities; there is little doubt that the viewer would use a few of them. The question is: *will* the viewer use them? The graphical user interface is the key. The on-screen graphics will either empower viewers to take full advantage of everything the technology makes available or frustrate them so they will not appreciate the power of the technology.

## The Challenge

Graphical user interfaces on televisions, VCRs, satellite receivers and cable terminals will be challenged to entice viewers to relax in front of the television set. Viewers will be inclined to spend more of their ever-decreasing spare time in front of the television set if the technology, and the graphical user interface that drives it, makes the experience fun and rewarding. Viewers will want to spend more disposable income on television programming and services -- both now and in the future -- if they feel the programming, the services, and the power of the technology enhance their family's home entertainment experience. Graphical user interfaces will play a crucial role in determining this feeling of satisfaction and value.

Graphical user interface systems being designed for consumer broadband video products will be a training ground for viewers. The assumption is that ever-increasing home computer sales or office or school computer use will prepare viewers for new interactive services or next-generation consumer broadband video equipment. This assumption may not be completely accurate. The level of confusion tolerated when operating a computer at home to reconcile one's finances, at school to finish one's term paper, or at work to create a slide presentation, may not be tolerated when trying to operate the electronic program guide to determine what is on television. This low tolerance level could be especially evident when one has limited choice in the video equipment or services offered by the broadband network service provider. Care must be taken to design graphical user interface systems that train viewers to interact differently with their television sets. A successfully designed graphical user interface will entice the viewer to use features and services that are complex without appearing to be. The learning curve should be shorter than that of learning how to operate most computer software. And over time, the viewer should not only feel confident navigating current service offerings, but also comfortable enough to subscribe to new ones when offered.

Potential revenue generating capabilities from new technology will depend on how comfortable viewers feel operating these initial systems. If the electronic program guide is frustrating to work, viewers will not use it. They will not pay for the service, and they could be disinclined to experiment further with other interactive services. If purchasing and watching a near video-on-demand (NVOD) movie is complicated, viewers will not purchase many NVOD events. Furthermore, they will be less likely to experiment with video-on-demand (VOD) services when they become available.

## Key Design Issues To Consider.

### *Design From the User's Perspective*

Designing a user interface from the user's point-of-view is easier said than done, but it is crucial in developing products that people can use intuitively. User-centered design means anticipating the user's expectations and questions then using design to address them. Instead of relying on explicit user manual directions to explain the product's capabilities, the designer relies on the design of the product's user interface itself as a guide through procedures. This design approach tends to be complicated because the designer becomes too familiar with the product's features and capabilities. It is easy for the designer to momentarily forget that the user does not have the same product knowledge. The designer must constantly ask himself: 'If I had never used this product, what would be my expectations of how it works? What are the design clues suggesting I do? Based on the patterns I have used so far, what would the user expect to do?'

A well-designed user interface does more than work with the current expectations of the customer. A successful user interface coaxes the user into trying unknown features. For example, most users would assume that an electronic program guide would tell what programs are on now and in the future and that the graphical user interface would help navigate through the guide. Users may not be aware that the program guide would allow them to sort and display programs by themes. The graphical user interface should suggest that this possibility exists and effortlessly guide them through the steps. Enticing and guiding users through the unknown in a stress-free fashion are the most important objectives of any graphical user interface. These are the paramount objectives of the graphical user interfaces being designed for consumer broadband television devices. How well these products can do this today could determine how willing viewers will be to use services tomorrow.

*Testing, Testing.*

The development of intuitive graphical user interface systems requires the early and continuous involvement of users throughout the development cycle. While focus groups, surveys and mall intercepts can be informative, the only real way to determine if a concept works is to test it on a typical user. Once the designer has an idea, a prototype is developed and given to someone who represents the type of person who would use this product. This person should not be familiar with the product or the interface. The subject, alone in a room with the prototype and the person conducting the test, is asked to complete certain tasks, but is not assisted in anyway by the tester. The tester watches how the subject interacts with the prototype. Ideally the subject would be videotaped and the tapes made available for study throughout the design process. Upon test completion the tester would interview the subject to determine why he approached tasks in that particular manner. The feedback from such testing tells the designer what elements work and do not work, and why. Videotape reviews allow the designer to learn helpful behavioral clues. This information is then incorporated into the design.

This type of testing can be used to analyze the effectiveness of a concept implementation or screen composition, or to choose between several implementation ideas. Unlike surveys and other types of market research, user testing sample size need not be large because the objective is to observe how the subject interacts with the design in order to improve it. One need not test 100 subjects if the first ten behave similarly. This is especially true if the sessions are videotaped because the tapes provide firsthand information. Once the design is complete, user testing should be conducted to determine if it is indeed easy to use. Feedback from testing can be incorporated into the design prior to release, or in the next release of products.

*Helpful Help.*

Even the most intuitive graphical user interface system needs to provide help instructions. Not everyone approaches unknown tasks the same way. Some people take chances when they reach a point of uncertainty, pressing buttons and dealing with the results. Others feel intimidated and will not proceed without seeking help first. Well-designed graphical user interface systems accommodate both groups of users.

Help can be categorized into two groups, general help, and specialized help. General help describes basic navigational and operational procedures. Specialized help is function-specific. While the basic navigational information may sufficiently execute a specific function, this may not, for some reason, be apparent to the user.

The risk-averse users must be able to access specialized help on a specific function and read that its execution is no different from other known functions. The graphical user interface designer is free to implement these types of help; the only requirement being that they are easily accessible.

*Mistakes Should Not Be Fatal.*

For those who are willing to take risks, mistakes cannot be traumatizing. Pressing the wrong key should not confuse the user. Incorrect buttons are pushed and wrong choices are made, both intentionally and by mistake. Choice confirmation and the ability to return to a previous screen or exit the graphical user interface system entirely are ways to restore user control when a mistake occurs.

*Remote Controls Are Equally Important*

Less is more; versus more is better. There are many schools of thought on the number of buttons that should be on the remote control. Actually, the most important thing is the ease with which a user can locate the buttons. The feel of the remote control in one's hand also is important. Is the remote control big and awkward? Are the buttons big enough given their importance to the user interface? Are the most frequently used buttons easily accessible? Are the labels easy to see in dim light? There are many, many styles and price ranges from which to choose. While price is a major factor in choosing a remote control, cost savings gained from choosing an inexpensive but difficult-to-use unit can be wiped out by the loss in service revenue because the remote control inhibits the user's access to services.

## Conclusion

Broadband television technology will provide viewers with more programming and service choices. Unfortunately, technological advances in other areas of life are leaving viewers less time and disposable income to spend on these new and exciting services. This increases the importance of the graphical user interfaces currently being developed for consumer devices. Graphical user interface systems designed from the user's point of view instead of the designer's point of view will be more successful in empowering users in navigating the many choices made available to them. The more comfortable viewers feel navigating the InfoBahn today, the more likely they will be willing to access it tomorrow.

# Managing the Return Spectrum to Optimize Interactive Revenue Opportunities

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## ABSTRACT

Issues related to reverse transmission in cable plant are considered. A topology is described which removes the largest portion of the interference in the return band. A few of the issues related to protocols suitable to return data are mentioned, and the idea of spectrum management is introduced.

## INTRODUCTION

For years the cable industry has had available technology that *almost* enabled the return band from 5 to 30 MHz. Some systems have employed the return band to transport video from local venues to the headend, and a very small number of systems have employed the return path to retrieve data from homes. However, little use has been made of the return spectrum until now. Cable operators have recognized that they will have to provide return for new services, however, and most systems now either have return path or can add it. In the days when systems were built according to tree and branch architectures (and there are a lot of such plants still in use), the return band was frequently considered impractical to use. However, with the modern fiber to the node architectures, the return band is usable.

A survey of several cable operators indicated expectation of revenue from services using reverse spectrum, ranging from \$3.00 per month (average for 100% of homes served) to over \$50.00 per month (in 25% of homes served). Revenue comes from such services as on line service (Prodigy, AOL, Internet, etc.), telephony (including long distance access) and interactive advertising.

## ISSUES IN SELECTION OF A FREQUENCY PLAN FOR UPSTREAM USE

Any attempt to set a frequency plan for upstream traffic must first take into account existing services if any. Most of the existing services are either video return from remote venues, or impulse pay per view systems with RF return. While we expect future services to be controlled with some sort of intelligent system, existing systems will continue to be controlled manually. Future control systems are discussed below.

Interfering carriers will also render some frequencies inoperable for all except perhaps very low data rate services. A review of the literature suggests that frequencies below 10 MHz are very difficult to use, and that frequencies above 20 MHz are much less susceptible to interference. A point of concern with the use of frequencies above 30 MHz, is the potential for interference to TVs and other devices connected to the cable. TVs in North America use the band from 41-47 MHz as their intermediate frequency. A typical level for a return signal leaving the home, is about 50 dBmV according to current thinking. This is as much as 50 dB above the incoming signal the TV is receiving. If a significant part of the upstream energy strikes the TV (due to limited coupler isolation), then interference is possible.

# History and Present Status of The Decoder Interface

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**Abstract:**

This paper reviews the decoder interface from its inception as Multiport, through the present development work to satisfy customer desires to use all the features available on their TV or VCR while maintaining all the functions and features available to the Cable Box user. The use and control of the TV/VCR tuner with a decoder interface, the interconnection between units, and the video, audio, and control bus techniques will be discussed. Of special interest is the section describing the control language which enables this system to work.

**History:**

Over the years there has been a love-hate relationship with the "Cable Converter". Often the converter overcame consumers' TV reception problems, while other times the converter was the gateway to receiving programming not otherwise available. While providing improved TV performance to the cable subscriber, most times only one channel at a time is presented to the TV for viewing. The desire to remove the converter, and its second remote control, prompted the idea of Multiport. Actually this was not a unique solution because Zenith had the "Redi-Plug" connector on Tvs during the same time frame. Basically it was suggested that the TV and VCR circuitry could be used if some control and descrambling were done external to the TV/VCR. The joint Electronics Industries Association (EIA) and the National Cable Television Association (NCTA) Engineering Committee worked to design an interface between the TV/VCR that the consumer owns, and the Cable Conditional Access device (descrambler) which the Cable supplier owns. The result of this was the first "Multiport", which was described in EIA Interim Specification 15 (IS-15). After some time, and the incorporation of some changes, the IS-15 specification was standardized as EIA-563. Among the changes was the incorporation of a one way command set to the decoder from the TV. This was because addressability was becoming popular and such control was deemed necessary. Prior to that time, scrambling was done on a "Pay" basis, a subscription type service, which required a truck roll for service level change. While a couple of million Televisions were made with Multiport Connectors, there was no way of knowing who had them in a cable system. The other problem was that Tvs from the largest manufacturer of Tvs with multiport did not work well with the most popular scrambling system decoders. General Instrument, and others, made IS-15 multiport descramblers, however, not too many were ever deployed. Few, if any, Tvs or descramblers meeting the full EIA-563 Multiport specification were ever sold.

## IS-105:

The emergence of Picture in Picture (PIP) Tvs obviously created another opportunity to improve the way Tvs and converters communicate. The PIP issue, on top of what the consumer felt were unfair costs for the products and services that they received from being connected to Cable TV, prompted the US Congress to enact the Cable Act of 1992. The law directs the Federal Communications Commission (FCC) to resolve all of these issues. The FCC has requested the Cable and Electronics Industries to present a solution to compatibility concerns and other issues. The FCC declared that, lacking a suitable solution, it will resolve these issues in their own way.

The joint EIA/NCTA has been working on a number of issues with the goal of creating systems which will supply good pictures at a reasonable cost. IS-23 looks to describe all of the factors involved in interfacing a TV, VCR or Converter to the cable system. IS-132 is a successor to IS-6 and attempts to standardize the channel assignments for analog signals to 800 Mhz. IS-105 is the successor to EIA-563, but very different. While the TV/VCR IF and detector still exist, their video output is not presented as an output. Instead, the TV/VCR tuner output at 45 Mhz is split with one path for the signal being the normal TV/VCR IF for detection and display, and the other path is to an "F" connector on the back of the TV/VCR. That signal is terminated in the first of a possible chain of devices using the IF of the TV/VCR. In every decoder box, a sample of the IF at unity gain is passed on to an output terminal for the next device. The interesting point here is that the TV/VCR delayed Automatic Gain Control (AGC) can be multiplexed back up the IF coaxial cable by the device actually using the tuners' IF. If the signal is in the clear, the TV/VCR will do the tuner AGC. Only if the signal is scrambled or uses digital modulation will the decoder need to do tuner AGC. The descrambler, or feature box, will supply baseband video, and audio if necessary, back to the TV/VCR through a multiconductor connector. The connector is the standard for the Audio-Video Bus (AVBus). This is a subset of the CEBus, created by the EIA for control in a whole house environment. The AVBus is targeted at connecting Tvs, VCRs, Disc Players, etc together without having to have a large number of cables running between the units. The minimum configuration of the AVBus for the decoder interface connector is to have pairs of wires, balanced for noise immunity; incorporating a ground reference, a control pair, one video pair, and one audio pair. The full configuration decoder interface adds three video pairs, and three audio pairs. With this many pairs, the video can be composite, or use two pairs for S-VHS connections, and audio can be carried as stereo pairs. The minimum configuration is for the TV/VCR to be "receive only" on the videos and audios, while the decoder will be "send only" on those pairs. In all cases, the control is "bi-directional" with a high degree of contention resolution capability. Multiple decoders for different scrambling systems, or to supply different features, can simultaneously operate to satisfy the consumers needs.

## Control Bus:

The control bus is probably the most complex of all the operations in the decoder interface. The data lines are normally relaxed, and everyone listens for an assertion by someone else. If there is no activity, and a device desires an operation, that device can take over using the data lines. At the same time that someone is using the control line they must also listen, in case a second user started at the same moment generating a contention. If that does occur, both users will stop, each will wait a different length of time and try again. Since the times to wait are different, the contention should be resolved.

## Control Bus Data Format:

The unasserted state is the normal off state. A "one" bit is 100 Microseconds ($\mu$sec) long while a "zero" bit is 200 $\mu$sec long. At the ending of a sequence, such as a recipient address, or a source address, an "End of Field" (EOF) bit lasting 300 $\mu$sec is sent. The end of a whole sequence is confirmed by sending an "End Of Packet" (EOP), data bit 400 $\mu$sec long. The data bits have only a time relationship, in that the very first bit goes assertive, whether it is a one or zero, and the second bit goes non-assertive etc. If the last bit in a packet is asserted, then the 400 $\mu$sec EOP would be non-asserted, and in reality a non-bit. The device sending the packet would normally wait for an acknowledge packet before starting a next packet. In most cases, the bus activity will not be so high that the next user would not wait to see if this was indeed the end of the packet.

## Data Packet Structure:

The data protocol is described in EIA IS-60 (the CEBus standard). There are two types of data frames likely to be on the Bus. The first is the Normal frame, which is used to send an original message, and the second is an Acknowledge frame, which is used to confirm receipt of a valid Normal frame. Before describing these frames in detail, it is worth mentioning that there are some commands which are "broadcast", in that there is no target address send and no Acknowledge is expected. These commands include, channel change and power on/off change. There may be more of this type of command included in the future, primarily used to shorten the time to send a command. In the Normal and Acknowledge frames, some of the fields are mandatory and some are optional. All fields must be accounted for however, so empty fields are identified by sending only an EOF bit. The PRE (preamble) field is defined as 8 random bits. The Control field is 8 Bits defined as X0000101. The X bit is a sequence number bit which toggles for each new message. The next field is for the DA (Destination Address). If sent as a null field, the packet is defined as broadcast to all devices on the bus. Address fields can be up to 16 bits long. The recommended practice is to try to keep the addresses to below 15. The DHC (Destination House Code) can be up to 16 bits long used to define the specific bus that is active. In a non-interconnected system, that can be a null field sent as an EOF bit. The next field is the mandatory SA (Source Address). This is followed by the SHC (Source House Code) field, which can be another null field, sending only EOF. Following all of the previous is the actual Information Field. This can be a field up to 32 bytes (8 bits of data per byte). The Information Field of a Normal Frame contains the Network and Application Layer headers preceding the Command to be executed. The actual command information is detailed in IS-105.2. The control possibilities are almost unlimited in that if the need arises for new commands, they can be added by mutual consent of the EIA and NCTA, and the specification revised accordingly. The final field in the sequence is the FCS (Frame Check Sum) which is actually the two's complement of the sum of all the fields on a byte by byte basis. The preamble is excluded in the calculation as are mathematical carries. The beauty of this scheme is that a register is cleared at the receiver during the reception of the preamble. After that, all the fields are added to the register on a byte by byte basis, including the FCS, which will result in the register being zero if no errors are detected. As was related earlier, an Acknowledge frame will be sent by the receiving device. The decoder can start to execute the received instruction at the same time that it is sending the Acknowledge Packet.

With the packet overhead and optimal short addresses, a Normal Packet can be as short as 8,800 $\mu$sec or 8.8 Milliseconds (ms). If the message is long, and addresses high etc, the packet may be as long as 26.9 ms. The Acknowledge Packet can be as short as 8.1 ms and as long as about 16 ms. This time has to be added to the time to transmit an IR command, which is typically 50 to 100 ms.

Upon the receipt of a typical channel change request, originated by the IR controller, a channel change command would be broadcast on the control bus. If the decoder recognized from its IF input that the channel was scrambled, a packet would be sent reporting that status. If the program could be descrambled, and if the decoder was not presently supplying video (and/or audio) through the decoder interface connector, the appropriate configuration would be requested by the decoder. The TV would respond, and a descrambled picture would appear on the TV. The IR transmit/receive time, which is not controlled, is added to the approximate 40 ms time that the Packets take going back and forth to establish the path. The time for the descrambler to recognize and authorize the channel might be another 16 ms. All of these times are based on smart assignment of addresses, and in that context, a reasonably short time elapses between the consumer request and the desired result.

## Conclusions:

This paper has presented the parts of a rather complex system. The complexity of the system should be weighed against the flexibility which it imparts. For example, the TV could tune to a scrambled channel, which the descrambler would process and present in a viewable form to the AVBus. That picture could be received and overlaid with data from another decoder box, before being presented to the TV for viewing. The time for packets to traverse the system is variable, and for the most part should be tolerable. The system is designed for setting up system configurations and not for playing interactive games. For fast reaction results, either IR Passthrough or a separate IR receiver may be necessary.

# Hybrid AM/16-VSB Lightwave Transmission System in Presence of Optical Reflections

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## Abstract

The performance characteristics of the 16-VSB cable modem was measured in a hybrid AM/16-VSB lightwave video transmission system using three different laser transmitters with the 16-VSB signal 6 dB below the AM carriers. The hybrid video lightwave system-based externally-modulated laser transmitters had ≤ 1.5 dB reflection-indepedent penalty due to impulse noise with a operating margin ≥ 9.1 dB for the 16-VSB signal even at high reflection levels (≥ -30 dB). In contrast, the hybrid video lightwave system-based directly-modulated DFB laser transmitter showed a 2.4 dB penalty due to impulse noise with a 9.6 dB operating margin at low reflections with no operating margin if the reflections are above -35 dB. However, for good performance of the AM channels, reflections less than -50 dB are needed.

## 1. INTRODUCTION

Hybrid multichannel AM/QAM (quadrature-amplitude modulation) or AM/VSB (vestigial sideband) subcarrier-multiplexed video lightwave transmission systems are attractive for video distribution and trunking.[1-3] Such systems can deliver existing broadcast multichannel analog video services while allowing the delivery of new compressed digital video and data services in the 500 MHz - 1GHz band. Recently, the 8/16-VSB modems were selected by the Grand Alliance as a transmission standard for terrestrial and cable broadcast of high-definition television (HDTV).[4] Currently, there is a technical discussion about the advantages and disadvantages of using 8/16-VSB versus 64/256-QAM (quadrature amplitude modulation) cable modem technology for digital video transmission over fiber/coax networks.[5] A digital cable modem standard would accelerate large scale deployment of digital set-top boxes and reduce future network cost.

In this paper, we report the first study of 16-VSB cable modem performance in a hybrid multichannel AM/16-VSB video lightwave transmission system. Impulse-induced clipping noise and optical reflections are known to cause degradations in hybrid AM/QAM video lightwave systems.[6] In a typical transmission link, there are many potential sources of optical reflections such as fiber optic splices, connectors, and couplers. Three different types of laser transmitters are compared: a directly-modulated (DM) 1310 nm DFB laser transmitter, an externally-modulated (EM) 1319 nm YAG laser transmitter, and an EM 1554 nm DFB laser transmitter. The performance of these laser transmitters are compared using (a) CW video carriers from a Matrix generator, and (b) a 50%/50% mixture of AM CATV video signals and CW video carriers from a Matrix generator.

## 2. 16-VSB CABLE MODEM DESIGN

The Zenith's 16-VSB cable modem operates at 43 Mb/s with SNR threshold values of 28 dB (in a 6 MHz bandwidth). The 16-VSB signal spectrum is flat throughout the band except at the band edges where it is roll-off with a 11.5% root-Nyquist filter, and a pilot carrier approximately 310 kHz above the lower channel edge.[4] Although the pilot carrier uses only 0.3 dB of the 16-VSB signal, it is essential for synchronous detection of the RF data signal and the symbol clock recovery. The 16-VSB modem receiver uses a symbol-spaced equalizer running at 10.7 MHz with a 63 tap real FIR filter. Forward error correction (FEC) code is employed to protect the transmitted data from burst of errors that might occur. The 16-VSB cable modem uses a T=10 (207,187) Reed-Solomon (R-S) forward-error correcting code (FEC). In our testing, T=10 (170,150) R-S FEC was used because of hardware availability. This modem is the same as the one tested by the Advisory Committee on Advance Television for HDTV.[4]

## 3. EXPERIMENTAL SET-UP

The experimental set up and the frequency allocation plan is shown in Fig. 1.

A 43 Mb/s 16-VSB signal centered at 537 MHz (channel 76) was subcarrier-multiplexed with 78 AM-VSB CATV signals (55.25 MHz to 547.25 MHz) and was used to modulate each of the laser transmitters. The RF power of the transmitted 16-VSB signal was 6 dB below that of the AM carriers. After transmission through two variable backreflectors (VBRs) and 4.4 km of standard single-mode fiber, the 16-VSB signal was detected at -1.8 dBm optical power, demodulated, and then fed to an error-detector. Better than 50 dB optical isolation was measured between the VBRs and each of the laser transmitters. A double-loop fiber polarization controller was used to maximize phase-to-intensity noise conversion (worst-case measurement). FC/APC SMF connectors, which typically have a return loss of less than -60 dB, were used in all measurements to minimize additional interferometric noise. In order to evaluate the impact of impairments on the 16-VSB modem performance with the R-S FEC, a white Gaussian noise (WGN) was added to the received 16-VSB signal to determine the required signal-to-noise ratio (SNR) threshold to achieve a bit-error-rate (BER) $\leq 3 \cdot 10^{-6}$. This BER has been defined as the threshold of visibility for the HDTV.[4]

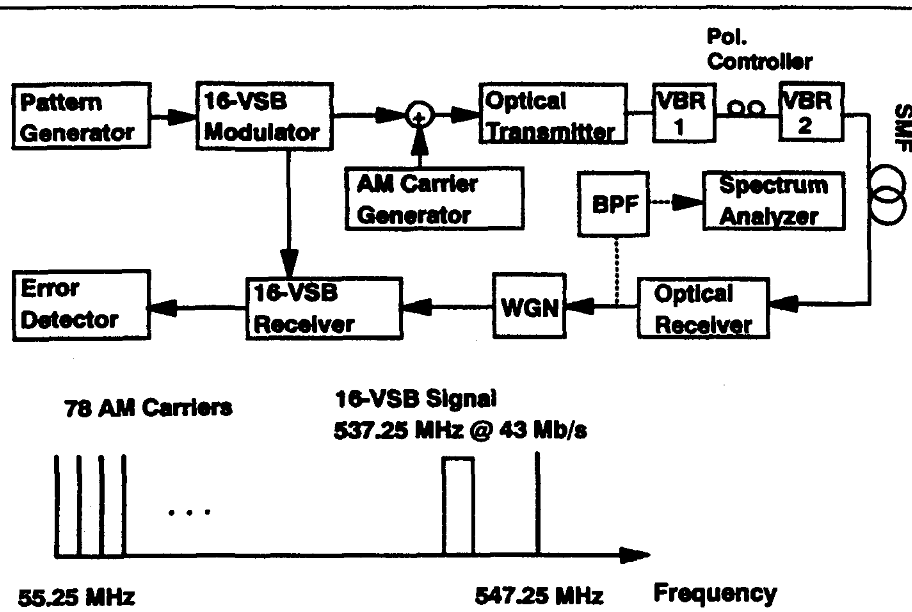


Fig. 1. The experimental setup and the frequency allocation spectrum.

## 4. RESULTS AND DISCUSSIONS

### 4.1. Directly-Modulated DFB laser Transmitters

The DM-DFB laser transmitter is the most widely used (> 80%) laser transmitter to transport the AM/digital video signals from the headend to the subscriber. The DM-DFB laser transmitter, which operated at wavelength of 1317 nm, had an optical power of 7 mW and low relative-intensity-noise (RIN ≅-159 dBc/Hz). Fig. 2 shows the

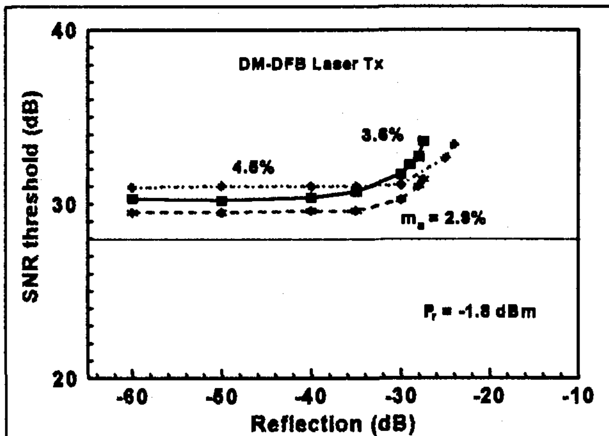


Fig. 2. The SNR threshold as a function of the reflection level for a DM-DFB laser transmitter with AM modulation indices of 2.9%, 3.6%, and 4.5% per channel using CW carriers from a Matrix generator.

SNR threshold as a function of the reflection level for the DM-DFB laser transmitter-based video lightwave system with AM modulation indices of 2.9%, 3.6%, and 4.5% per channel using CW carriers from a Matrix generator. The measured analog CNR was 48 dB at an AM modulation index of 3.6% per channel with the CSO and CTB distortions less than -65 dBc. With no WGN added, the received 16-VSB SNR was 40 dB. At a low reflection level (< -50 dB), the 16-VSB SNR threshold is increased by 2.4 dB from the reference SNR threshold level of 28 dB (the back to back case) for an AM modulation index of 3.6% per channel. Thus, one has a 9.6 dB operating margin to threshold at low reflection level, which is the difference between the received 16-VSB SNR and the SNR threshold. The 16-VSB SNR threshold is rapidly increased at reflection levels greater than -40 dB. At reflection levels greater than -27.5 dB, the R-S FEC was overwhelmed and the measured BER was greater than $3 \cdot 10^{-6}$ even with no added WGN. This means that the 16-VSB SNR threshold is higher than the received 16-VSB SNR, and thus the 16-VSB modem would not operate with the DM-DFB laser transmitter in this case. On the other hand, if one keeps low reflection levels and increase the AM modulation index to 4.5% per channel, the 16-VSB SNR operating margin is reduced to 8.8 dB, representing occurrence of additional errors due to increased clipping-induced impulse noise.[3]

Fig. 3 shows the 16-VSB SNR threshold as

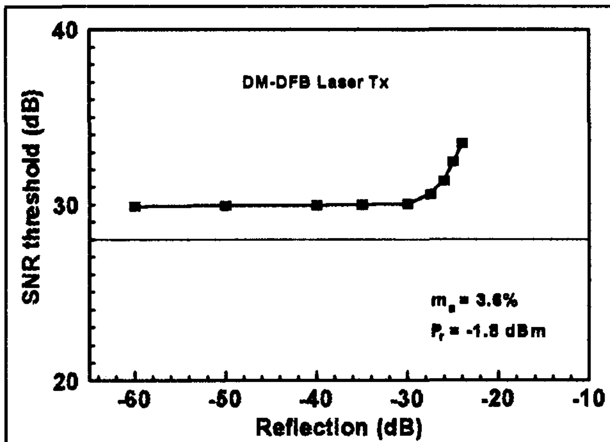


Fig. 3. The 16-VSB SNR threshold as a function of the reflection level with the same AM modulation index as in Fig. 2, but using a 50%/50% mixture of AM CATV video signals and CW carriers.

a function of the reflection level with the same 3.6% AM modulation index as in Fig. 2, but using a 50%/50% mixture of AM CATV video signals and CW carriers from a Matrix generator. Notice that the 16-VSB SNR threshold is reduced by 0.5 dB compared with the results in Fig. 3 at low reflection levels, and no 16-VSB operating margin for optical reflections greater than -24 dB compared to -27.5 dB (Fig. 2). Changing the AM modulation index per channel for the DFB laser transmitter-based lightwave system at a reflection level of -60 dB causes no significant degradation in the observed 16-VSB SNR threshold as shown in Fig. 4. In contrast, the SNR threshold is rapidly degraded when only CW carriers from a Matrix generator are used (Fig.4). Thus, when using a 50%/50% mixture of

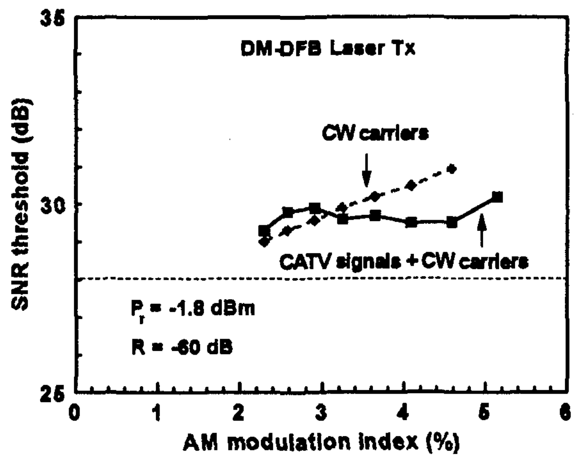


Fig.4. The 16-VSB SNR threshold as a function of the AM modulation index per channel for a DM-DFB laser transmitter using a 50%/50% mixture of AM CATV video signals and CW carriers.

AM CATV video signals and CW video carriers from a Matrix generator, the reflection tolerance is increased by up to 10 dB compared with CW video carriers. In practice, reflection levels less than -35 dB are required for robust transmission of only the 16-VSB signals. However, the AM CATV signals require reflection levels less than -50 dB in such a system.[6]

## 4.2 Externally-Modulated Laser Transmitters

The EM-YAG laser transmitter ($\lambda$=1319 nm) has two output ports with 20 mW at each port, and a very low RIN (RIN≈-170 dB/Hz).[6] The EM-DFB laser transmitter, which is an attractive alternative to the directly-modulated (DM) DFB laser

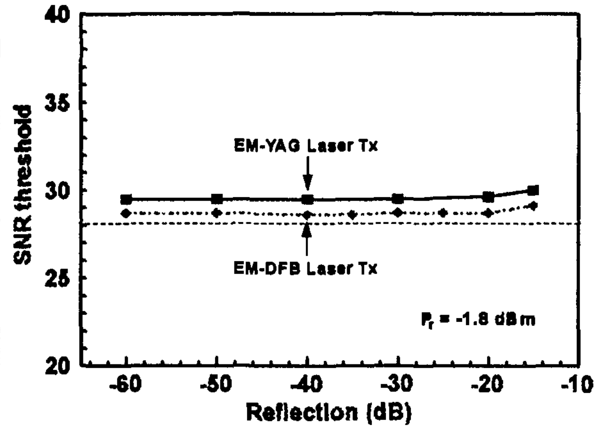


Fig. 5. 16-VSB SNR threshold as a function of the reflection level for the EM-YAG laser transmitter and the EM-DFB laser transmitter.

transmitter, operated at 1554 nm and incorporated an Er-doped optical fiber amplifier with a 20 mW output optical power and a low RIN (RIN ≈ -162 dB/Hz).[7]
Figure 5 shows the 16-VSB SNR threshold as a function of the reflection level for the EM-YAG laser transmitter (AM modulation index of 3.1%) and the EM-DFB laser transmitter (AM modulation index of 3.2%). The analog CNR was 48 dB with the CSO and CTB distortions less than -65 dBc. The received 16-VSB SNR was 40 dB. The 16-VSB SNR threshold had a reflection-indepedent 1.5 dB penalty due to impulse noise and a 10.5 dB operating margin using the EM-YAG laser transmitter. Similarly, the 16-VSB SNR threshold had a reflection-indepedent 0.8 dB penalty due to impulse noise with a 11.2 dB operating margin at low optical reflections using the EM-DFB laser transmitter. However, at a high reflection level (> -20 dB), the operating margin is reduced by about 2 dB for this hybrid system.

The optical transport results can be explained as follows:[7] Interferometric intensity noise (IIN) is generated from two sources, namely, multiple discrete optical reflections between the two VBRs, and from the fiber's Rayleigh backscattering interacting with those reflections. The EM-YAG laser transmitter and the EM-DFB laser transmitter have a narrow linewidth (≤ 1 MHz) and a very low IIN ($RIN_t$ < -160 dB/Hz at R = -15 dB) so the observed 16-VSB SNR threshold is reflection-independent. In contrast, the DM-DFB laser transmitter has a relatively broad modulated linewidth (> 1 GHz), and therefore has significantly higher IIN ($RIN_t$ = -130 dB at R = -15 dB). Thus, large amount of optical reflections (> -40 dB) degrade the digital channel further.[5]

## 5. SUMMARY

In conclusion, the performance characteristics of the 16-VSB cable modem was measured in a hybrid AM/16-VSB lightwave video transmission system using three different laser transmitters with the 16-VSB signal 6 dB below the AM carriers. The hybrid video lightwave transmission system-based EM-YAG laser transmitter or EM-DFB laser transmitter had ≤ 1.5 dB penalty due to impulse noise with a reflection-indepedent operating margin ≥ 9.1 dB for the 16-VSB signal even at high reflection levels (≥ -20 dB). In contrast, the hybrid video lightwave system-based DM-DFB laser transmitter showed a 2.4 dB penalty due to impulse noise with a 9.6 dB operating margin at low reflection levels for the 16-VSB signal. From the measurement using a 50%/50% mixture of AM CATV video signals and CW video carriers from a Matrix generator, it is observed that the reflection tolerance is increased by up to 10 dB when actual modulated video signals are transmitted instead of CW carriers. In practice, reflection levels less than -35 dB are required for robust transmission of only the 16-VSB signals. However, the AM CATV signals in such a system require reflection levels less than -50 dB.

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# HYBRID FIBER-COAX NETWORK POWERING ISSUES

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## ABSTRACT

One of the most significant and controversial issues facing HFC network designers today is powering. With the deployment of HFC based networks by Telephony providers, the powering systems are evolving into a much different design than previously used by the CATV industry. Issues such as providing power down the drop cable to the home, network interface device power characteristics and power passing taps are important considerations. Other concerns for these new designs include choice of operating voltage and frequency, corrosion, safety, current limiting, NEC and NESC code compliance as well as the unique requirements of Broadband telephony. This paper will review the powering issues specific to the distribution area with emphasis on powering requirements for Broadband Telephony on HFC networks.

## INTRODUCTION

Telephone, cable television, and utility operating companies have begun to deploy new full service broadband networks. This upgrade of the information infrastructure will require new approaches to powering and power protection of valuable electronics. Networks currently under strongest consideration include the hybrid-fiber coaxial architecture using asynchronous transfer mode (ATM) electronics. Distributed line card technologies are also a key approach. For these architectures, major issues with respect to equipment protection and safety will be explored. A pertinent body of knowledge has been acquired through years of cable television, computer, and telephone industry use of related equipment when exposed to their respective fault hazard environments. New networks will bring together a broad range of technologies and fault hazards, but at the same time must meet even higher levels of reliability and service availability. Recommendations for circuit protection will be offered, and a summary of efforts being made today on setting standards will be presented.

## NETWORK EVOLUTION

Today's cable television and telephone industries are on the verge of massive changes in both how they conduct business and the physical plant that they use. Although traditional telephone service will continue to play an important role in telephone company business, the greatest growth potential is in expanded services, including home entertainment programming. For cable companies, expansion of services will also be desirable, resulting in a convergence of offerings and demand for networking technologies. While telephone companies have an excellent reputation for maintaining their service, their physical plant with its miles of twisted pair copper wire is in many respects obsolete Recognizing this, many operating companies are embarking on huge projects to deploy fiber-coaxial distribution networks.

In upgrading to new networks, companies will be introducing equipment such as ATM network interface cards, Telephony and Data Modems into much more electrically abusive environments than that of traditional LAN applications.

For reliability and performance, ideally all of the existing copper would be replaced with optical fiber. However, bringing fiber to the home presents the telephone companies with several problems that they would rather not address at present. Chief among them is how to interface the optical cable to the electronic equipment being used by the customer. Equipping each customer line with an electro-optic transceiver is very expensive. There is also the question of how to provide power to the customer premises for both transceivers and equipment (Network interface Devices and phones).

## PRACTICAL IMPLEMENTATIONS OF FIBER IN THE LOOP: HYBRID-FIBER COAX

Rather than bring the optical fiber inside the subscriber's premises, telephone companies are now thinking of bringing fiber to the curb or further back from the home. In the case of fiber to the curb, the optical fiber is brought to a remote terminal, or optical network unit (ONU), which is connected to customer premises equipment by copper wire, and to video services equipment via coaxial cable. These conductors carry electrical power to remote telephone company equipment and the customer's equipment. Although there are several variations on this theme, industry sentiment appears to favor a hybrid-fiber coaxial architecture. This approach provides a high-quality broadband connection to the home that rivals optical fiber over short hauls, but is much less expensive.

As now envisioned, optical fiber serves as a trunk between the central office/headend and a remote node that contains the opto-electronic translation equipment. Coaxial cables connect this mode directly or through intermediate amplifiers to network interface units located on the outside walls of customers' premises. In a typical proposed implementation, a coaxial distribution cable from the node is hung on utility poles (aerial plant application). This cable carries both the signal information and local operating power in the form of 60VAC or 90VAC, 60 Hz Quasi-square wave. Coaxial taps provide the means through which four or eight subscriber lines can be connected to the coaxial distribution cable One approach is to carry the two-way RF signal over a coaxial cable to the network interface unit (NIU), and to provide power for the NIU and customer premises equipment over a separate pair of copper wires. This approach is attractive to existing Telephony providers that already have the twisted pair drop to the home that can effectively be re-used now as a power conductor. This approach can also be justified for deployment due to concerns of corrosion and high resistance of the aluminum outer conductor of the coaxial drop cable and F-connector interfaces in use today for non-power carrying CATV service.

Another major concern is the RF insertion loss increase and performance degradation possible in power passing taps that will require a power diplexer circuit, current limit circuit and power inserter function in each tap output port. At each network interface device, another power diplexer is required to isolate the power signal from RF for use in the NIU power supply module.

Aside from the use of optical fiber and coaxial cable, the biggest difference between the new and old approach to telephone networks is the location of the subscriber line interface card (SLIC).

The SLIC, which contains the switching relays that connect a customer's equipment to a ring voltage generator and to the network, is traditionally located in the central office or more recently, a remote terminal. However, in a fiber-in-the-loop implementation, the SLIC is moved closer to the customer. Two architectures are envisioned; In one, the coaxial tap on the pole or in a street-level pedestal contains the SLIC. As a result, power is passed directly to the phoneset and the NIU is passive. In the other approach, taken by most of the hybrid-fiber coaxial (HFC) networks being considered today, each NIU contains its own dedicated SLIC, local ring generator and loop power for the subscriber phone devices. Regardless of location, the operating power output requirements of the SLIC station must be compatible with the existing installed base of phones. The worst case architecture from a powering point of view requires an NIU for each subscriber premises location. The power overhead requirements of each SLIC, ring generator and Switchmode Power Supply (SMPS) add an estimated **40 to 60%** increase in total power consumption for a 480 subscriber node. This power consumption is **in addition** to the power required by the fiber node and amplifiers required for a stand-alone CATV system.

The inclusion of the SLIC inside each NIU or curb site significantly increases the total amount of power being consumed by downstream operating equipment. Moreover, when broadband programming and service become available, the customer premises equipment needed to support it will incorporate sophisticated microcontroller based circuitry as well as RF manipulation functions such as interdiction, addressablity, self diagnostics and more. Depending on the design, some or all of this equipment will draw electrical power from the telephone line. As a result, operators providing telephony in this manner face increased power supply output power requirements as well as higher through-currents for active and passive devices in series. As a result, the higher fault currents possible (as well as higher voltages) significantly increases the potential for catastrophe if an electrical fault develops anywhere in the copper or coax portion of the downstream network.

## **Problems in the power path.**

The transmission media used by most local ATM networks today are located in a benign environment -- inside cable tunnels in a campus-wide backbone, for example. However, the situation is quite different in a typical telephone distribution network. Consider, for example, a traditional copper-based telephone system. Calls are routed from one telephone to another through switching circuits over fiber and the copper wire transmission system, much of which is above ground.

These aerial plants are exposed for miles to a variety of environmental hazards. The most dramatic is that of a pole from which a telephone line is suspended being struck and knocked over by a vehicle. If the pole is also carrying power lines, such an event could cause the telephone and power lines to come into direct contact (Power cross), which could place hundreds or even thousands of volts on the telephone line. Less dramatic but more commonplace than these so-called "power crosses" is the induction of high voltage transient spikes on a telephone line due to nearby lightning strikes and utility neutral return currents. Although they may be of short duration, these voltage transients can achieve peak amplitudes of several thousand volts.

Both power crosses and induced transients can have devastating effects on network and customer premises equipment. Of even greater importance, a power cross can be fatal to telephone company personnel working on the line. Under certain circumstances, it also can be fatal to a telephone user.

## TRANSIENTS, CURRENT LIMIT ISSUES

To minimize the effect of power crosses and lightning-induced transients, as well as downstream short circuits, telephone operating companies employ a number of voltage and current limiters. Overvoltage protection is typically provided by gas discharge tubes or solid-state transient clipping devices such as Avalanche diodes, MOV's etc. - connecting each line to ground. Voltage is lowered across the load when overvoltage devices are triggered, creating a much lower resistance to ground (gas tube) or through dissipation of energy (metal oxide varistor). Current limiting is typically performed by thermal type devices that go into an "open" state when heated with $I^2R$ internal dissipation. Current limiting is performed by heat coils, thermal resistors, and other fuse-like devices connected in series with each line. In a traditional telephone system, most of these limiters are located in the central office. Because central offices are often staffed around the clock, replacement of a blown fuse or heat coil could be performed within a matter of minutes at a moderate cost.

Implementation of fiber-in-the-loop architecture moves the focus of protection out of the central office and into the field. Even in copper-based systems, the telephone network is rapidly changing to accommodate changes in population and expansion of commercial zones from center city areas to outlying districts. As part of this change, some of the functions traditionally performed in central offices are being moved to remote terminals adjacent to areas with heavy telephone use. Whether the remote terminals are part of an all-copper network or a fiber/copper network, they contain sensitive switching circuits that must be protected against power crosses, induction, and transient voltage spikes. Of equal importance, the customer premises equipment also must be protected.

Remote equipment can be protected by the same kind of voltage and current limiters used in central offices. However, the use of heat coils and other fuse-like current limiters in remote terminals may not be pain free. If a power cross or voltage transient with sufficient energy occurs, these fuse-like current limiters will blow to protect the equipment in the remote terminal. The problem is that these remote terminals are not staffed, which means a technician must be dispatched to the terminal to replace the blown current limiters. Not only is this expensive, it results in a loss of telephone services to a potentially large number of subscribers for a significant period of time.

When fiber-in-the-loop architecture (e.g., FTTC and HFC) is implemented, the potential for power crosses or the induction of high voltage transients between the central office and remote node virtually disappears. However, there is still a significant potential for power crosses between the node and the Network Interface Unit (NIU) if the copper distribution cable is hung on power poles, as is often the case.

If a hybrid-fiber coax architecture is being used, the distribution cable is coaxial. Because the coaxial cable's outer conductor (sheath)is bonded to Utility neutral and ground periodically, lightning induced voltage transients can affect the network. In fact, Cable TV systems that have "Over grounded" their plant have actually increased the problem by reducing the coax plant apparent impedance to utility neutral imbalance currents thus increasing current flow through the sheath and strand ! A study covering two Southeastern cable TV systems revealed a number of instances in which currents were induced in the grounded shield as a result of nearby lightning strikes and resultant utility imbalance.

A significant issue is the transient voltage protection approach used for coaxial distribution verses the previously described copper twisted pair. Solid state protection devices cannot be placed directly across the coax center conductor to sheath due to the parasitic capacitance and inductance that will cause unacceptable RF degradation. Use of Gas tube devices has been discouraged in the CATV industry due to the AC power supply voltage present on the network that provides a follow-on current when a Gas tube triggers. Over time, the gas tube will fail in a conductive state causing network failure. "Crowbar devices" have been proven effective for use in transient protection but can only be used on the RF isolated side of a power inserter or power diplexer in an amplifier.

Because the potential for a power cross or induced high voltage transient is real, especially in broadband systems, telephone operating companies must provide adequate protection.

Cable Television Companies are currently deploying broadband telephony. The SCTE-Society of cable television Engineers and the NCTA-National Cable Television Association as well as Cablelabs are CATV oriented organizations working towards recommending broadband telephony solutions. Various telephone standards and specifications promulgated by Bellcore, ITU-T (formerly CCITT), and other similar organizations have dealt with copper systems (although this is changing rapidly). These can be applied to the copper portion of the hybrid-fiber coaxial system and other fiber-in-the-loop architectures. However, the distribution of fault energy onto the outer conductor of coax must be better understood before more exacting recommended practices are drafted. Key provisions of pertinent specifications are shown in reference Table following this article. The table shows points of protection, potentially applicable standards, and recommended practices.

None of the Bellcore and ITU-T network standards and specifications address systems in which the SLIC and associated power consuming electronics are attached to the wall of a customer's premises. Instead, the portion of the system that exists from the coaxial tap to the customer's premises has been interpreted by many to be covered in whole or in part by the National Electrical Code (NEC). Section 800 of the NEC covers telephone lines along with the AC powered fire and burglar alarms. According to Section 800, a listed protector shall be provided on each circuit run partly or entirely in aerial wire or cable not confined within a block. Also, a listed protector shall be provided on each circuit, aerial or underground so located as to be exposed to over 300 volts to ground. Listed protectors to date, however, are not functional for use with coaxial cable RF systems.

Another key provision of the NEC is in section 725 which suggests that a protected circuit be limited so that the total available power during a fault near premises wiring does not exceed 100 VA. While the NEC provides overall limits regarding the power capacity of telephone and CATV lines, they do not specifically address the lines that run from the NIU to the equipment inside the customer's premises. As a result, the CATV and telephone operating companies are applying those standards and specifications they believe are applicable, most notably Underwriter's Laboratories UL 497 and 497A. Even so, the question of which standards and specifications really apply remains unresolved and controversial. Circuit protection devices should encompass potential specification applicability, and, in addition, strive for resettability in order to maintain minimize down time.

## POWER NODES

Due to the combined power consumption of the broadband devices and the telephony related interface equipment, the size and complexity of the network power source has changed. Typical HFC architectures feature a centrally located fiber optic transceiver (node) with four coaxial feeder cables leaving the node in four directions to serve the four "quadrants" of the node. These feeders carry the two way RF signal as well as to conduct power from a power source located at the node to the amplifiers in series with the feeders to each quadrant distribution area. CATV only systems can just barely conduct sufficient power from the node to the active devices. Coax cable and connector resistance causes voltage drop and power loss and depending upon amplifier power consumption, coax size and feeder distance, quadrant power transmission is very limited. The most important factor is the through-current limitation of the power passing inductors and connectors integral to the amplifiers and passives. These devices have a fundamental through-current limitation of 12-15 amps with acceptable RF performance from MHz to 1GHZ. CATV HFC networks require 2400 to 3600 watts for the actives and transmission losses for an approximate 500 home node. Using 60 volt powering, up to 15 amps is required through each feeder from the central power source. This is at the upper limit of the power passing capability of the broadband devices. Parallel feeders could be employed from the node to each quadrant to double the power transmission capacity. This approach is not very attractive due to the extra construction cost and network operating cost. Another option is to increase the power system voltage. (As of this writing, several RBOC's have committed to use either 60 VAC or 90VAC, 60hz). At 90VAC, using the same example, only 10 amps is required to be conducted through each feeder. This is within the existing device power passing capability. All is not well though ! Our example only considers the power consumption of CATV actives and transmission losses. When the Telephony device load is added, we need all of that extra current and more (15-20amps!) even at 90 volts. Typical 500 home nodes with central powering require 40% more power for curb side telephony devices that provide loop power and ring voltage for say, 12-24 homes. Due to sharing of the power overhead for the loop power and ring generator by so many lines, overall node power required is lower. The side of the home NIU approach requires power to be transmitted to the furthest extremities of the network, the home! This means that power is routed through a power passing tap, down a coax or twisted pair drop to the interface device at the subscriber premises. Each of these devices contains the SLIC "line card" function, ring generator and loop power supply. This extra power consumption and loss can result in 500 home nodes requiring 5-7 kilowatts of power each. In addition, these power nodes are physically large, often include battery banks, integral generators and other features for redundancy.

## PROPOSED OVERVOLTAGE AND OVERCURRENT PROTECTION METHODS.

Because each NIU or coaxial tap would contain its own dedicated SLIC, each would also have to contain its own overvoltage and overcurrent protection devices. This means that if a power cross occurs or high energy transient appears on the line, the protection devices in dozens or even hundreds of NIUs or taps could be affected. If fuse-like current limiters are being used, it could mean that dozens or even hundreds of customers would lose their telephone service until a technician could replace all of the affected protection modules.

Not only would this result in considerable loss of customer good will, it would represent a significant expense to the telephone operating company. Clearly, if broadband networks are to be implemented on a large scale, traditional fuse-like current limiters will have to be replaced by devices that automatically reset themselves after the fault clears. Not only to meet the code required 100 VA fault power limit, but to ensure that device damage is limited and undamaged devices reset automatically.

Not only is this clear for transient voltage events but for short circuit conditions such as caused by pinched drop cables, NIU faults, etc. The new power nodes can deliver 2 kilowatts or more down each feeder prior to current limiting. This is significant energy to deliver into a fault! Operators are very wary of the liability issues for field staff as well as for customer premises faults. Often, the coax drop is stapled to the outside of the home and faces significant heating and potentially, combustion if the current limit circuit in the tap or feeder is not effective. Another consequence of drop fault current is the load on the adjacent tap output ports that causes loss of service to other subscribers - potentially all that are connected to the affected feeder. A method of current limiting in multiple network locations is required to reduce the fault affected areas to the minimum number of subscribers.
A proposed schematic of the drop is shown in the reference section, showing resettable fuses in addition to the traditionally resettable overvoltage protection.

One resettable device that has proven useful in other applications is the conductive polymer positive temperature coefficient (PTC) resistor . This device acts like a resettable fuse. Unlike conventional PTC resistors, which more linearly simply track temperature, conductive polymer devices provide the "snap action" associated with fuse-like devices. The key to their operation is the existence of conductive chains running through the nonconductive polymer body.

At normal operating temperatures, the polymer exhibits a crystalline structure. However, when the device's temperature rises above a critical level, its structure changes to an amorphous state and, as a result, the conductive chains break. This rapidly increases the device's resistance by several decades. Although this increase in resistance reduces the current flowing through it, the remaining current is sufficient to maintain the device in its amorphous state. This, in effect, latches the device into its tripped condition. When power is removed and the fault corrected, the device cools and its structure reverts to its crystalline state, restoring the conductive chains, and thereby reducing the resistance to its normal values.

This, in effect, resets the device. Use of these devices to limit current at remote locations provides protection for fault conditions and have been used effectively in existing SLIC twisted pair applications.

Additional approaches to current limiting also include active semiconductor based circuits such as a “smart circuit breaker” using relay contacts or thyristors for the interrupting function. A pulse width modulation or phase angle control can provide immediate disconnection or a constant current mode. These devices add cost, complexity and size to the tap and other locations such as the amplifier feeder outputs. Although accurate, and consistent over ambient temperature range, There is concern about the reliability of solid state devices like these exposed to the previously described transient voltage and energy discharge conditions.

Network operators are considering a distributed current limit system similar to utility power distribution. Circuit breakers and solid state current limiters are used in the power node outputs. The amplifiers would also employ individual current limit circuits in each of the feeder outputs of a lower threshold. A standalone device in series with the feeder could provide additional current limiting with an operator adjustable current limit set point. Each power passing tap would also require the PTC or active limiting options as previously described for each of the output ports. Further current limiting is required in the NIU to limit fault current that could be conducted into the home by the NIU SMPS. As can be seen, the highest current limit set-point is at the power source and each limiting device in series to the home employs a decreased let-through current threshold. In this manner, faults are limited to a reduced area by isolation of the current limit or disconnect function.

To summarize, the increased network power demand caused by the combination of broadband and telephony devices has resulted in the need for higher voltage and power output power sources. These centralized power nodes are required to economically provide power to the most area with the least number of power nodes required. This saves the operator not only the capital cost but the ongoing maintenance cost for batteries, generators and other internal components. With a minimum back-up time of 8 hours for lifeline telephony, these power nodes require a combination of battery banks for the UPS function as well as generators for economical long term back-up operation during AC power outages. Use of a power node for less than 500 subscribers further increases capital cost if the same 8 hour back-up requirements remain. Another consideration is the sheer number of these power sources required. Operators are concerned that in a disaster situation there would not be enough personnel available with generator or inverter equipped vehicles to provide back-up power to numerous power node sites. CATV operators are approaching this from the direction of currently using 3-5 900 watt power supplies per node and perhaps reducing the number to 2 or 3 with higher output voltage (90VAC)and 1.5 to 2.5 kW output power for equivalent reach to the actives but with reduced power supply locations to reduce capital cost. This approach still does not provide increased power margin for the added telephony device load. When this occurs, the single large wattage power node for 400-500 subscribers seems to reach the economic equilibrium point.

In addition to implementation of higher AC voltages, some projects have looked at DC powering or low frequency AC powering such as 1 cycle per second. There is a valid concern about safety with 60 volts although the CATV industry has used this standard for 15-20 years without apparent problems. At 90 volts, there is more of an issue. There is research data that shows increased heart fibrillation danger at frequencies above 10 hertz. So some have drawn the conclusion that a lower frequency of operation would trade the increased safety provided by the lower frequency for the increased risk of the higher operating voltage. Too low of a frequency or DC still is questionable due to the corrosion issues posed by the galvanic junctions of the cable and connectors etc. Plant reliability could be a problem at lower power frequencies and definitely would be with DC powering. At this time, there is not enough field experience to draw a fair conclusion. One effect of the different powering frequency is the increased cost and complexity of the power conversion devices in the power node. Full time inverter modules are required to provide the output power and redundant devices are also employed. This precludes the use of line frequency transformers and power circuits that have proven MTBF superiority in the CATV industry experience over the last 30 years.

As of this writing, The second largest MSO has started to implement 90 VAC, 60 Hz powering and several other MSO's are considering this. Three RBOC's have started deployment using 60VAC, 60hz power systems and two are planning on using 90VAC, 60HZ. One RBOC is deploying a 90V, Hz power system. In either Hz or 60hz, 90 volt systems, safety is a major concern. Amplifiers, taps and other devices need to improve shielding of fuses, conductors and voltage access points to improve safety. Improved warning labels and training will be required. There is concern about the use of the plastic ATO automotive type fuses in the amplifiers at the 90 volt level. In addition to the potential rating limitations of these fuses, the interest in self resetting limiters on feeder outputs for network reliability will most likely prevail.

## POWER PASSING TAP

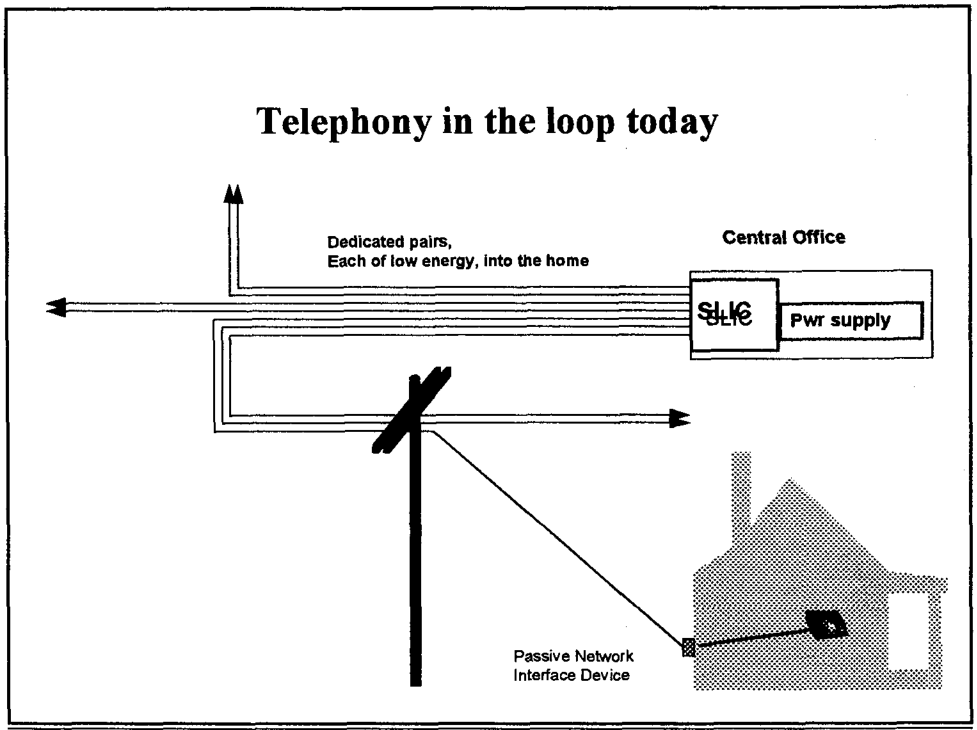
As discussed, the power passing tap requires an RF bypass network to provide power only to the current limit devices. The output side of the current limit circuit is fed to the output connector in the case of the twisted pair power drop option.

The coax drop power output version requires a power insertion function in series with the each output port to re-insert the current limited power to the RF output. Another power isolation network is required in the NIU to remove power only and route it to the SMPS and ring voltage generator. Thus the twisted pair output adds one additional RF insertion loss penalty compared to a standard tap where the coax power output version adds three insertion loss penalties. This is a concern because even if the operator is attracted to the simplicity of one drop to the home that carries both power and RF, the loss penalty of this approach could exceed the slim signal level margin that may exist and force the use of an additional amplifier in cascade. This then "ripples" through the design reducing performance and requiring even more power for the extra actives. Another required feature for both tap designs is a power and signal bypass function. If the tap cover plate is removed for service of a single port failure etc. all other downstream subscribers *loose signal and power.* Although this is irritating for subscribers of CATV only, it is unacceptable for networks carrying lifeline telephony. The next generation power passing taps will require a bypass function to avoid disconnect of downstream subscribers.

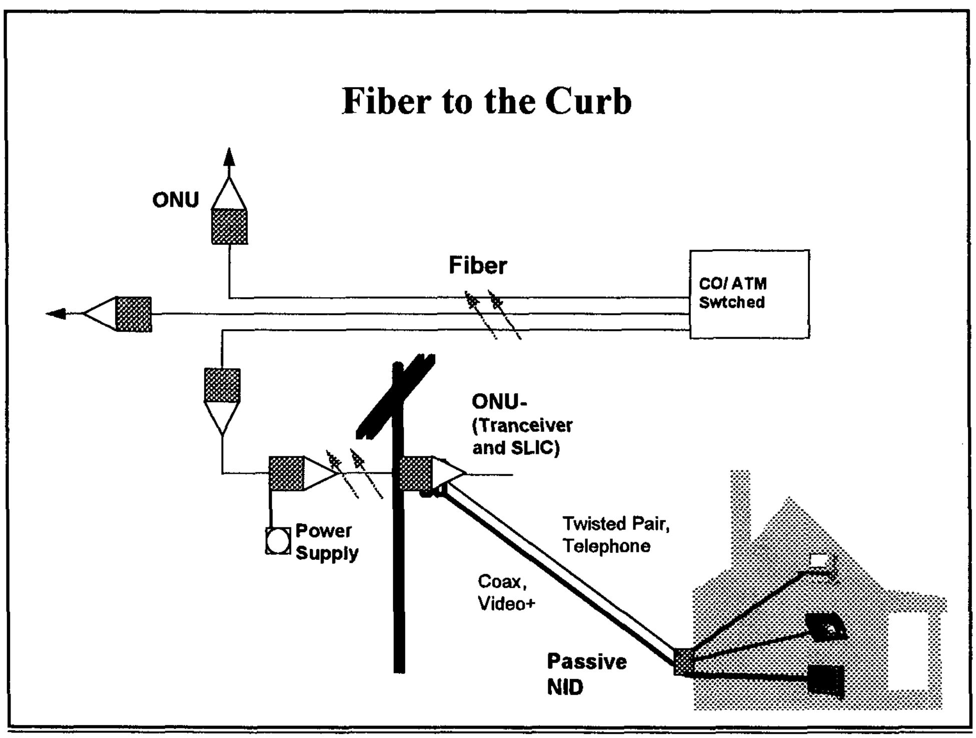
## SUMMARY

HFC Networks that carry telephony require significantly different approaches to power source size and configuration as well *power architecture*. Increased power consumption of the additional devices has forced consideration and use of higher voltages. Higher energy outputs of the power nodes has in turn focused designers on the current limit and fault protection functions from the power node to the telephone. In addition to code compliance, safety and reliability issues have come to the forefront in both network and equipment design. Several NIU manufacturers are working on reduction of power consumption, "sleep mode" functions etc. to reduce power load. Further marketing requirements for more sophisticated NIU signal functionality may counteract these power savings though. Also under consideration is subscriber power units with battery back-up in each network unit. Although outside of the scope of this article, this approach may prove feasible especially for wireless telephony designs and may prove reliable and functional while reducing operator cost compared to network powering. This is too early to confirm.

All network designers need to focus on the choice of the powering infrastructure including transmission issues, current and voltage limiting, safety and reliability of the design.

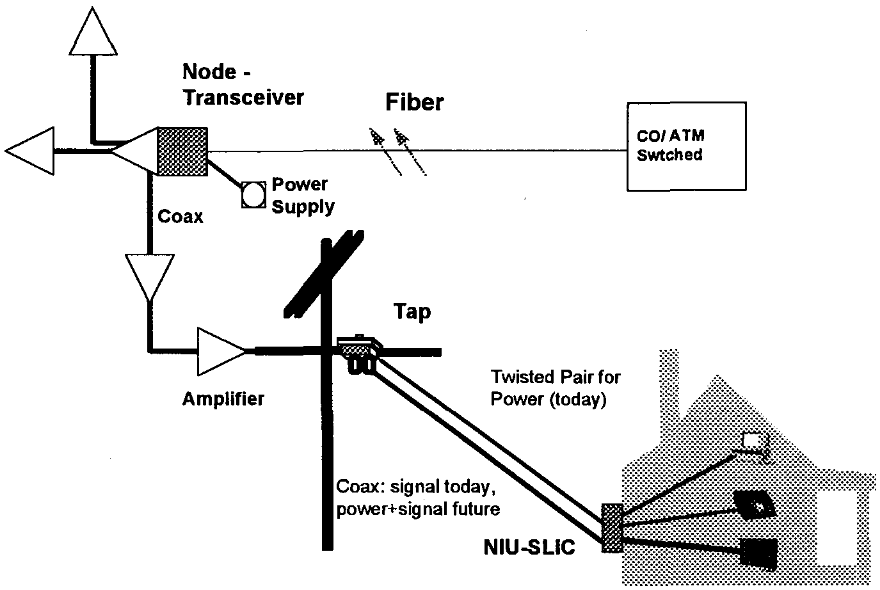


Telephony in the loop today
Dedicated pairs,
Each of low energy, into the home
Central Office
SLIC
Pwr supply
Passive Network
Interface Device

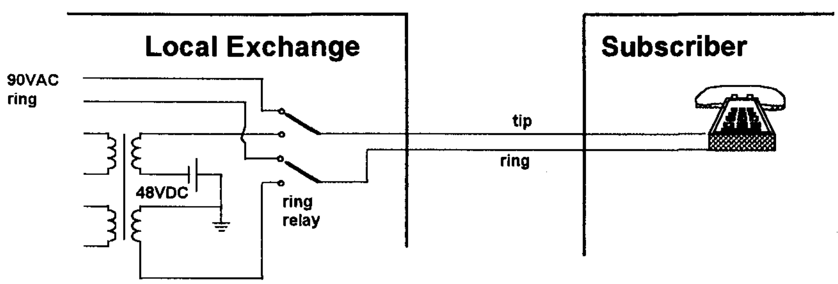



Fiber to the Curb
ONU
Fiber
CO/ ATM
Swtched
ONU-
(Tranceiver
and SLIC)
Power
Supply
Twisted Pair,
Telephone
Coax,
Video+
Passive
NID

## Hybrid Fiber-Coax




## Telephone Powering

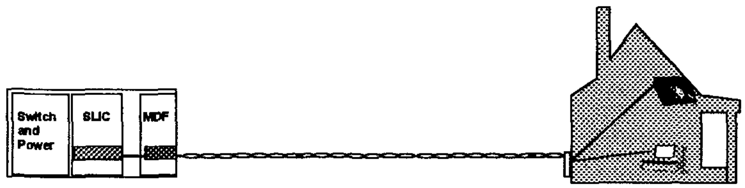



**'Local Exchange' location:**
**- Traditionally: Central Office**
**- HFC: Side of home (NIU)**
**- FTTC: At the curb (ONU)**

| | |
|---|---|
| **System Voltage** | **- 48 VDC** |
| **Ringing Voltage** | **90V rms, 20Hz** |
| **Operating Current** | **20 mA** |
| **Max. Current** | **70 mA** |

## Traditional Telephone Conditions

- Traditional Hazards:
- Post Lightning effects: High voltage, short duration
- AC Power contact: High energy, varying voltage and duration
- AC Power induction: Variable




Dedicated Power, SLIC to Phoneset:

| | |
|---|---|
| System Voltage | - 48 VDC |
| Ringing Voltage | 90V AC rms, 20Hz |
| Operating Current | 20 mA |
| Max. Current | 70 mA |

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# Intelligent User Interface Design for Interactive Television Applications

Steven Schlossstein
Interactive Health Network, Inc. -- HealthNet™

## 1.0 Introduction

A generation ago, when mainframe computing was the workplace standard, a high priesthood of software engineers was necessary for programming functions, system maintenance needs, and general interface with users. In the late 1970s, when the personal computer was born, computer hardware underwent a decentralizing process as the digital revolution started shifting power to the individual. By the 1980s, application software had become the driving force of the PC industry as the realization grew that personal computers were simply lifeless boxes without popular software that could run a myriad applications, from word processing to presentation graphics to spreadsheets.

Early software applications for the PC were controlled primarily by arcane, keyboard-driven, text-based commands; this was the generation of the Apple-II, the first IBM PC, the Kaypro, the Osborne, the TRS-80, and other popular platforms. They were soon supplanted by PC systems that incorporated a "friendlier" user interface; first, the familiar Macintosh, and then Microsoft's Windows operating system that brought to the screen a graphical design based on a desktop metaphor using windows, icons, a mouse, and pull-down menus (the familiar WIMP factor). But not all software applications designed with a graphical user interface, or GUI, operated in the same way, and users had to cope with (and still do) an often complicated functionality that requires keyboard or function-key input. Moreover, while software applications have become "easier" to use, they are still relatively "dumb;" that is, they do not learn from user experience or user preference, and often cannot adjust or adapt to user styles. They may more aptly be termed "system interfaces" than "user interfaces."

Now the digital revolution is creating a powerful force called convergence, in which computing and television technologies are coming together to create new systems, new applications, and new markets. The interactive television industry (ITV) is on the threshold of widespread system deployment; by late 1995, dozens of major ITV testbeds and trials will be underway in the continental U.S. Efforts by cable TV providers (Major Systems Operators, or MSOs) and the regional Bell operating companies (RBOCs, or telcos) will dramatically alter the ways in which people use and watch television, bringing greater control, choice, and convenience to the consumer.

Utilizing powerful video server hardware, asynchronous transfer mode (ATM) transmission, and MPEG-encoded digital video compression, these new systems will create dynamic new services such as movies-on-demand, electronic banking, home shopping, and interactive games. But user interfaces designed for entertainment applications may be inadequate to support enhanced interactive video services such health and medical information, education, distance learning, or job training.

These ITV systems may at first be extraordinarily difficult to navigate by the average consumer. They will necessitate the storage of immense quantities of compressed digital video data in remote databases on servers at the system's head-end. Utilizing a standard NTSC television receiver with an addressable set-top box, they will have no keyboards, mice, pull-down menus, or other familiar input devices. The consumer must be able to navigate the new ITV systems using only a hand-held, "smart" infra-red remote-control. And at a normal viewing distance of 10 feet, NTSC screen resolution precludes the use of typical windows or computer icons that have become so familiar in a desktop environment where the user sits one-on-one 18 inches from the monitor. Latency is seldom addressed as a problem in the PC environment where users may tolerate an on-screen hourglass endlessly as they wait for the system to carry out its complex functions, but it is a critical problem for ITV. Viewers will insist on instantaneous system response to their commands.

Navigating a small video library of popular, first-run movies may seem undaunting on the surface, but when that collection expands to consist of 10,000 titles ranging from classics to comedy, new and more intelligent video database navigation tools will become necessary. Yet even that problem pales by comparison to a large video database of enhanced video information services that may consist of an even larger collection of shorter video segments that necessitate interaction *within* the segments themselves. And this navigation issue becomes more critical still when we realize that the user is limited to a single command device (the remote control), which creates a special challenge for user interface design: how to display information on the screen (and what information to display), how to organize that information most effectively for the user, and how to incorporate intelligent agents in the user interface design that can learn from user preference and experience.

The Interactive Health Network, Inc., (HealthNet) is responding to this special challenge by developing an application that utilizes intelligent user interface design for enhanced video information services. Together with our technical alliances at Microsoft and the Cognetics Corporation, we have been working for the past year on appropriate user interface metaphors for enhanced video applications; researching mental models for con- sumers, including zero-trial learning with appropriate user testing through usability groups; experimenting with different user interface designs (including but not limited to text-only via attractive on-screen menus, TV icons ("bugs") only, icons with text, video-in-windows, and spatial navigators);configuring systems with minimal external controls; and conducting routine analysis of the relevance of video databases containing large numbers of short segments to the underlying user interface designs.

The system prototype developed by HealthNet and completed in late 1994 is intended to deliver critical health care information efficiently in video format to consumer and patient alike, which can greatly increase the quality of care provided and dramatically reduce burdensome tasks for the health care professional while lowering the overall cost of care. In institutional use, HealthNet's unique interactive video system can enable patients to access vital medical and health information to see pre-operative briefings, post-op therapies, and treatment protocols as well as video presentations on diet, nutrition, medication, fitness, and exercise. It has been estimated that nearly half of all doctor visits may be related to issues that do not require the costly attention of a skilled healthcare professional. A major benefit to the nation is that HealthNet's new interactive video system will support and enhance clinical pathways critical to both higher health care quality and lower health care costs. At the same time, it will contribute to innovative, high-skill job creation in the emerging interactive television industry. And it can help reduce the U.S. trade deficit by showing how America's high-tech service applications could be exported to ITV systems developers in other countries, notably western Europe and East Asia.

## 2.0 Background and Technical Approach

There are several forces at work today that require a new look at how user interface designs are created and deployed. One is the sheer explosion in software availability overall, driven in part by the continued decline in the cost of microprocessor power; as PC hardware becomes cheaper and more broadly available, so do the software applications designed to run on it. Functionality is increasingly controlled by ordinary end users, many (if not most) of whom even feel overwhelmed by the relatively simple interface demands of the VCR, which, despite numerous attempts to correct its design flaws, still blinks "12:00" in most homes. As hardware has become cheaper, it has also become more complex. Hewlett-Packard's first generation of laserjet printers, for example, contained about 25,000 lines of code. Today, they have ten times that amount, and the new models expected next year may contain a million lines of code. Users need more (and better) help learning how to use (and master) these products. Specialists in interface design are being challenged to do for new-generation systems like ITV what the telegraph did for messaging a century ago when it forced the obsolescence of the Pony Express. The simple migration of existing designs from computing to television will not be sufficient to do the job.

The digital revolution has also created virtually endless growth in communication services, especially for data. (Since mid-1992, AT&T's long-distance service has been dominated by data, not voice, transmission.) This in turn has dramatically expanded the number (and diversity) of people who use computers, software, modems, and online services in their daily work. As these communications technologies continue to grow and expand, so will the number of general users. This puts increasing pressure on user interface specialists to create simpler, more comfortable, easier-to-use designs that can make digital services more accessible to the general user. Nor is this a frivolous challenge: creating simple "up-front" designs that mask complex, intelligent "back-end" tasks implies a sea-change in attitudes, because designers can no longer afford to assume that users are primarily other computer-literate professionals.

The devices with which users use to interface with systems have also multiplied in recent years, leading to new efforts in speech recognition, gesture recognition, data-gloves, head-mounted displays, touch-screens, and virtual environments. Most of these are at best imperfect devices today, still in need of huge improvement before they can become accepted in the consumer marketplace; many -- the virtual reality interfaces -- further restrict system usage or benefit to one user at a time, which conflicts with the more open (and shared) television environment. And they all suffer from one major flaw, which is that they tend to be designed by experts for use by other experts, with very little attention to the general, non-technical user. As a result, the new challenge is to develop *user-centered* interface designs.

This challenge is all the more daunting for the successful and efficient navigation of enhanced video services such as medical and health information, distance learning, or job training. With the development and deployment of more effective, innovative, and intelligent user-centered interface designs, interactive television systems that distribute medical and health information can enable universal access to high-quality health care information so that the nation's best specialists will be available to rural residents and inner city patients alike.

HealthNet has focused its research specifically on user-centered interface design. Given the absence of traditional computer-based devices (such as windows, icons, mice, and pull-down menus) in a video environment, this approach has been predicated on the need for a simple, hand-held, remote-control device. HealthNet's work to date has researched several innovative, attractive, and easy-to-use "front-end" user interface designs that mask a more complex "back-end" intelligent functionality. We have tried working with several designs that incorporate these important elements, focusing in particular on two interface designs, one based on a conversational metaphor, the other on an "intelligent agent" metaphor,

In cooperation with Cognetics Corporation, HealthNet has worked on research, design, and demonstration through frequent usability testing of an innovative user interface and de minimis hand-held remote device that have optimum applicability to enhanced interactive video services such as medical and health information. HealthNet's technical approach has been structured around a multi-step process based on the QUE Design Methodology™ developed at Cognetics to facilitate the integration of usability engineering into new software designs, along the following lines:

◆ Develop the interface concept.

◆ Identify the functionality of each interface model.

◆ Identify the user population, an issue that is critical to effective usability engineering design.

◆ Identify high-level constraints, such as hardware architecture and system software compatibility.

◆ Create a storyboard sketch for each interface metaphor in development.

◆ Identify the usability goals for the interface.

◆ Plan the screen layout.

◆ Analyze the input device and design appropriate action sequences.

◆ Develop screen prototypes built around key screens that reflect the conceptual models, the tasks they will accomplish, and their look and feel.

◆ Iterate the designs and refine them, using a rapid prototyping tool like Visual Basic v.3.0 for Windows, iterated over four cycles:

◆ Final review and usability testing to determine an optimal interface design and de minimis remote control device.

## 3.0 HEALTHNET'S INTELLIGENT USER INTERFACE DESIGN FOR ENHANCED VIDEO SERVICES

HealthNet's Advisory Board suggests that it may be possible to cover a majority of common-interest health and medical subjects in about two-dozen basic video segments. We have been working to refine those choices, cross-reference the categories through keyword menus, and analyze other ways of representing the appropriate paths to them -- work that will be enhanced in future research.

But the navigation of a video database consisting of enhanced video services is orders of magnitude different from simply selecting a movie in a video-on-demand service or buying a product in a home shopping application. It is not necessarily more difficult, depending upon the user interface design, but it is clearly different.

Access to HealthNet's interactive video segments is enabled by means of drop-down "bugs" that are called onto the screen while the viewer is watching normal broadcast TV. [This effect is easily achieved through a function key on the remote control.] These bugs, or icons, represent interactive services available on the viewer's system and might symbolize movies, games, shopping, sports, news, network reruns, education or reference services, financial markets, and health. When a viewer highlights and selects the HealthNet icon, he or she is switched immediately into the HealthNet video database and remains there until making a clear "exit" decision.

We are working on input of user profiles [name, age, height, weight, and brief medical history] that the system can retain based on which particular family member may be using the system. However, because most IR remote controls do not contain alpha keys and viewers are typically not comfortable using a remote control for data entry purposes, data input of this nature must be calibrated to the de minimus remote control and enabled by means of either numeric entries (age, height, weight) or entries in a fixed field ("1" equals yes, "0" means no).

HealthNet's main screen opens with video running in the background -- typically a talk show format with well known guests discussing current or controversial issues related to medicine and health. The HealthNet Navigator, consisting of large square icons arrayed horizontally across the bottom of the screen, is overlayed on top of the running video, with a generic Topics button in the lower right-hand corner and an Exit button in the lower left. The four Navigator icons graphically depict the principal demographic categories -- children, women, men, and seniors -- from which a viewer may choose to access the underlying subject (and video) of choice.

Highlighting and selecting the Women's icon, for example -- simply by pushing the appropriate direction arrows and then the "Select" key on the remote control -- takes the viewer into the Women's screen, which consists of a menu overlay containing six keywords centered vertically from top to bottom and easily readable from the normal viewing distance of 10-12 feet. Through an initial period of tests and trials, HealthNet has discovered the truism that one keyword is worth a thousand icons.

The six choices available on the Women's screen represent three areas of general interest (Updates, Fitness, and Nutrition) and three diseases or disorders specific to this demographic category (Breast Cancer, Menopause, and Stress). While this keyword list is clearly not exhaustive, it does represent convenient access to subjects of immediate interest to women. [A more complete list of choices is discussed under the Topics option, below.]

Highlighting and selecting the Nutrition option pops open a listbox showing a full range of keyword choices such as healthy snacks, high-fiber diets, low-cholesterol meals, low-fat meals, vitamins, etc, in alphabetical order. Selecting one of these options then opens a data form into which the viewer can enter salient statistics (such as age, height, and weight) if these have not been pre-determined earlier.

Highlighting and selecting the "OK" button on the data screen starts the video clip on the chosen subject -- high-fiber diets, for example -- with an 3-5 minute overview of the topic. As soon as any HealthNet segment begins, a separate overlay menu appears containing two sole options -- "Exit" to the lower left, "More" to the lower right. Selecting "Exit" immediately returns the viewer to the access screen -- in this case, the Women's screen -- if a mistake was made or if the viewer is done.

Highlighting and selecting "More" brings up a final overlay menu containing several keywords representing additional choices the viewer can make in order to interact within the video segment itself. For Nutrition, these choices are Recipes, Snacks, Quick Meals, Menus, Books, and Services. Selecting any one of these stops the current video segment and starts the chosen video, always with the "Exit" and "More" options at the bototm of the screen.

HealthNet has taken a similar approach in designing the user interface for video segments related to specific diseases or disorders. For example, from the Women's screen the viewer may choose "Breast Cancer," and be switched immeditely to that segment. From the Exit/More screen, selecting "More" brings up the overlay menu with six keyword choices, all while the overview segment is playing in the background.

These six keyword choices are Causes, Symptoms, Complications, Prevention, Treatment, and Services. Highlighting and selecting any one of these keywords starts the video segment relevant to that particular choice, thus enabling the viewer to navigate (or browse) the HealthNet content library quite easily.

HealthNet is currently working on an additional layer of material so that the viewer may be able to "drill down" one layer deeper into each of these six areas and watch more detailed video presentations on each of them. Under "Breast Cancer," for example, once the viewer is in the "Treatment" segment, he or she may wish to access more specific segments that relate to primary treatment protocols such as Lumpectomy, Mastectomy, Mastectomy with Reconstruction, or Watchful Waiting. Each of these choices is again arrayed as a keyword option on the screen.

Highlighting and selecting the "Topics" button in the lower right-hand corner of the main HealthNet Navigator screen brings up another overlay menu containing six general keywords -- Nutrition, Fitness, Symptoms, Diseases, Injuries, and Services. Hightlighting and selecting any one of them pops open a listbox containing an alphabetical list of detailed choices relevant to that keyword selection.

Keywords are important in navigating health and medical information, we feel, because they most closely depict a conversational metaphor. Picking movies from a poster-like array of choices may be fine for entertainment, with catchy music and alluring previews built-in. Or having the screen resemble a colorful direct-mail catalogue like L. L. Bean or Land's End could work well for a clothing outlet. But access to important health and medical information in our view requires a "quiet corner" of the interactive services menu in order for consumers to benefit from and use an enhanced service like HealthNet effectively.

HealthNet has examined and surveyed a number of proposed user interface designs for other interactive services, such as movies on demand, interactive games, home shopping, and financial markets. Some of these user interface designs are attractive, innovative, and clever. But they are also daunting to the first-time user, and imply a familiarity with computers and pointing devices that the ordinary television viewer may not have.

"Video in Windows" is one popular example. Highlighting and selecting a choice on screen pops open a small window in which a video clip begins to play. Again, as a preview of full-length Hollywood movies this U/I approach is not without merit. But a screen containing four videos playing simultaneously in adjacent windows, as one design is currently structured, can be confusing to both the eye and the ear.

The combination of icons and text is another mistake often made when creating new user interface designs for interactive television. If the icons are too small, they are virtually unrecognizable on an NTSC monitor because of the poor resolution on the one hand and the greater viewing distance on the other. HealthNet has used only four oversized icons in its Navigator because they symbolize recognizable demographic categories -- children, women, men, seniors -- and sized at about 3" square, they are easily visible from a distance. But large icons also take up a lot of space on the screen, making it virtually impossible to use them together with text.

Text works best when used as keywords -- again, HealthNet's experience is that one keyword is worth a thousand icons. But too much text crammed onto a single screen makes it illegible for the same reasons that icons are hard to read -- poor resolution and greater viewing distance. In our experience, there is still much work that needs to be done in refining the keyword approach to video navigation, and our initial goal of keeping intermediate screens to a minimum may have to be revised. From the main screen to an underlying video segment, the viewer is never more than three layers removed from the targeted segment.

A final problem that will need attention is the multiplicity of different navigators planned for interactive video services around the country. While there have been frequent calls for a navigation standard to be developed so that the differences between system navigators can be minimized, we think it will be many years before one emerges -- just as it took more than a decade for Windows to become the accepted U/I for desktop computing. HealthNet is continuing to research these various ITV user issues.

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# INTERNET ACCESS VIA CABLE TV: HIGH SPEED ACCESS TO THE INFORMATION HIGHWAY

**Lynn Jones**
**Digital Equipment Corporation**

*Abstract*

***Internet access has become a feeding frenzy. Dial-up access to the World Wide Web just isn't fast enough. Your subscribers know that cable TV can give them access to hypertext at the speed of a bullet train, and they want it. Now what do you do?***

***How do you separate the hype from the hypertext? For that matter, what is hypertext? This paper will answer these questions and more, providing a framework within which to plan and implement Internet access on your system. It will give you a crash course on the Internet and show how the information highway is being created today with the Internet and cable TV.***

## INTRODUCTION

*"Internet over cable TV will receive more general press (Information Superhighway) attention than any other carrier-based activity in 1995 and become the most important tool for work-at-home since the advent of the PC, modem and facsimile terminal"*

> *Dr. Jerome Lucas*
> *TeleStrategies Insight Newsletter*
> *January 1995*

The commercialization of the Internet has opened the door for access to the largest information network in the world. The Internet connects an estimated 39K networks, 4.9M hosts[i], and over 30M users in more than 127 countries around the world. In early 1994, it was estimated that on the average, a new computer was added to the Internet every 30 seconds.[ii] The growth of the Internet continues to be exponential.

The cable TV industry has the capability to provide subscribers with Internet access at speeds unmatched in price/performance by any other medium. Using the ubiquitous, existing Hybrid Fiber Coax (HFC) infrastructure in place in the US, Canada, and many countries around the world today, the cable industry can give customers high speed access at a much lower cost - a key criteria for Internet users.

Rapid access to and provision of pictures, sound, video, audio, and integrated text and graphics are all made possible by accessing Internet sites on the World Wide Web, using a Web browser program from a personal computer connected to the Internet via cable television. By using standard channels, one forward and one reverse, on an existing entertainment cable system, cable operators can offer businesses, telecommuters, doctors, teachers, students, and consumers access the state-of-the-art capabilities available on information highway today.

This paper discusses creating such networks using an Ethernet to cable TV bridge, brouter, or personal modem at each business, hospital, school, or home, coupled with a translator in the Cable TV headend and a backbone router to the Internet at one location on the network. Each user on the network has access to the Internet, as well a shared 10 Mbps access to each other. The value of the cable connection to the customer is increased by orders of magnitude.

The key is the ability of cable to provide a fully distributed network, rather than customers using expensive point-to-point leased lines for direct connection to the Internet, or slow dial-up lines for indirect connection through online services.

The goals of this paper are to explain the network technology, components, and architecture, and the methodology which will provide the cable operator with a foundation on which to plan and build such a network. Case studies of actual networks are used to illustrate possible configurations.

The convergence of the cable and data communications industries provides enormous opportunities, but carries with it the challenges of learning about each others' technologies. This paper is intended to serve as a starting point, assuming little or no knowledge of data communications or the Internet.

## A BRIEF HISTORY OF THE INTERNET

The evolution of cable TV data networks can be seen to parallel the development and growth of the Internet. Both have the roots of their technology growing largely out of military applications with initial use and promotion of the technology by the education and research community, followed by adoption in the commercial marketplace, and finally reaching widespread use in the home.

A brief history of the Internet helps to understand why it is organized the way it is, and how cable networks connect to it.

In the mid-1970's, the Defense Advanced Research Project Agency (DARPA) funded research to develop a set of networking standards or protocols, that specify how computers would communicate over an internet, as well as a series of conventions for interconnecting networks and routing traffic. The result was TCP/IP (Transmission Control Protocol/Internet Protocol). During the late 1970's, DARPA also funded research into *packet switched* networking and implemented a network called the ARPAnet.

TCP/IP became the only effective way to communicate between computers from different manufacturers. It appealed to schools, institutions, and businesses, who did not want to be tied to one vendor's equipment, and who wanted to protect their investment in existing equipment.

In the early 1980's Ethernet local area networks (LANs) proliferated. Ethernet, developed by Metcalfe and Boggs in 1976, used a coaxial cable network, in which all stations monitor the cable (the ether) during their own transmission, terminating transmission immediately if a collision is detected. This created a new demand: rather than connecting to a single large timesharing computer per site, organizations wanted to connect the ARPAnet to their entire local network. This would allow all the computers on that LAN to access ARPANET facilities.

In 1986, the National Science Foundation founded NSFNET to connect its networks centered around its 6 supercomputers into a network *backbone* that ties into the ARPANET.

This network arrangement was enormously successful, and the Internet was born. With success came the need to upgrade compute resources and leased line speeds, which continues today.

## MANAGEMENT OF THE INTERNET

The Internet has evolved from a loose federation of networks, to a network with a character all its own. There is no central management, but rather a group of organizations who steer its activities. These groups include the Internet Architecture Board (IAB), the Internet Society (ISOC), and the Internet Engineering Task Force (IETF).

Another unique aspect of the Internet is that, as it is not owned by any one party, it is also not paid for or funded by any one organization. NSF, which subsidized its development is phasing out its $11.5M subsidy. Privatization of the Internet remains a hot topic. Many third party Internet access providers have sprung up to offer Internet access to businesses and individuals.

## THE COMMERCIALIZATION OF THE INTERNET

Those who stand to gain most from the commercialization of the Internet are small businesses, K-12 schools, home workers, and recreational users. Corporations, research institutions, and universities have been using the Internet for many years. Big business has also been using private *wide area networks (WANs)* for years as well. (WANs being essentially company-owned internets.)

Prior to the commercialization of the Internet, small businesses, schools, and individuals could not afford the high price of private WANs, and were not allowed access to the Internet. Commercialization opens the door for access to resources previously only available to large organizations. It also opens up a whole new range of possibilities. Virtual corporations and electronic shopping malls are not only possible, but possible on an international level. Internet marketing and advertising will change the way products are promoted.

The commercialization of the Internet will forever change the future of both small business and worldwide commerce. It will also forever change what the Internet is.

## WHO ARE THE USERS?

Users of the Internet include companies, universities, colleges, K-12 schools, research groups, and individual users.

The majority of universities and research facilities have Internet access. Many companies also have access. There is now a major initiative to connect K-12 schools. Individual users are largely an untapped market, as until recently, acquiring Internet access required specialized knowledge of *UNIX* and TCP/IP.

The popular press is rife with articles on the wealth of information available on the Internet, as well as on the need for a national, as well as global, Information Highway. This has sparked a feeding frenzy of interest in connecting up to the Internet. To people today, the Internet *is* the Information Highway.

## WHAT ARE THE APPLICATIONS?

Applications on the Internet started out with simple text-based applications, such as electronic mail. Today's applications are highly visual - containing color pictures, sound, graphics, video, and other data-intensive information formats. It is the increase in such applications that drives the need for the "big pipes" that cable TV can provide.

## Electronic Mail

The most commonly used application on the Internet is *electronic mail*, or *e-mail.* Each user has a unique address and can be reached by anyone else with email access to the Internet.

Electronic mail is a low-bandwidth, text-based application. It is a *store and forward* service, meaning it does not require communicating users to have an *end to end* communications path set up at the time the message is sent. The message is forwarded and stored on subsequent computers until is reaches the recipient.

Traditional low-bandwidth, store and forward applications such as e-mail, do not necessarily require the high-bandwidth capabilities of a cable TV network.

## The World Wide Web

The *World Wide Web,* or *Web* is the newest and most often talked about application on the Internet today.

The World Wide Web provides easily accessible, organized access to the huge amount of data available on the Internet. The Web uses *hypertext,* in which displayed information contains highlighted words which can be "expanded" with the click of a mouse, providing links to other information or files. These links can be to text, pictures, video clips, audio clips, graphics, etc.

The Web is a *client-server* based application. Information databases are stored on computers called *Web servers.* In order to access the Web, a user's computer needs a piece of software installed called a *client*, or in hypertext terminology, a *browser.*

Multimedia, client-server based applications like the Web are the future of the Internet, and fuel the need for cable TV speeds. It is becoming commonplace for small business, schools, and municipalities to have their own Web servers. This trend necessitates high-speed *symmetrical* access, meaning that access speed is the same in either direction, which cable TV is in a unique position to provide.

## Wide Area Information Servers (WAIS)

WAIS provides a means to search indexed material using a string of text supplied by the requester. It allows the user to easily look for information regardless of where it is located on the Internet. WAIS is one of many such search tools, which instead of browsing randomly, allow a user to locate specific information.

## Gopher

*Gopher* allows a user to *tunnel* through the Internet and access information without having to know its address. Using gopher is like having a library card catalog to access information, rather than having to search for it randomly.

## News

*News* allows users to access information on a variety of topics or special interests, and is analogous to a discussion group or bulletin board.

## File Transfer Protocol (FTP)

The File Transfer Protocol (ftp) is a client-server based application used for copying files from one computer to another over the Internet. Ftp sets up a *real-time* connection between the two computers while the copying is taking place.

## Telnet

*Telnet* is another client-server application that allows a user to log in to another computer on the Internet. It can allow users to access databases, public information, and library card catalogs.

## Summary

One of the primary lamentations of Internet users is the inability to get high speed access to applications such as the World Wide Web from their desktop (particularly from home) where a dial-up line may be the only available option. A dial-up line may be fine for e-mail, but it is totally unsuitable for applications such as the Web. The time it takes to download an image is just too long for the average user to endure. Symmetrical access is also key, as small companies, schools, and home-based businesses set up their own Web servers.

## THE CASE FOR INTERNET ACCESS VIA CABLE TV

*"...sometimes it makes sense to back out of the driveway at 900 miles per hour."*

*Vint Cerf*
*co-author of TCP/IP/Internet designer*

Communications mirrors society, as well as changes the way society interacts. Both data networks and cable television evolved to bring information to people in physically distant locations. People and networks have become more decentralized. An outgrowth of the ability to communicate over wide distances has fueled the growth of virtual companies, telecommuting, distance learning, and other phenomena. Key to these models are the fact that the people using networks can be either producers or consumers of information, or *prosumers*. This societal model, coupled with the need to access the data-intensive applications used in these activities requires high-speed symmetrical network access.

Key to the commercial growth is the ability to provide users in remote locations, at small companies, and at home with the same high speed access to information both on community networks and on the Internet that they are used to having in the workplace.

As Vint Cerf, one of the founding fathers of the Internet, points out, "The information superhighway model, with low-speed access to a high-speed backbone, is flawed. My experience with data networking is that sometimes it makes sense to back out of the driveway at 900 miles per hour. We need to support both low-speed and high-speed access. For that reason, narrowband, 128-kb/s integrated services digital network connections are not bad, but developments like cable TV-provided 10-Mbps Ethernet links are even more interesting."[iii]

Standard telephony lines simply do not have the capacity to bring the same bandwidth to the home as cable TV.

## HOW DOES INTERNET ACCESS VIA CABLE TV WORK?

Data networks use devices such as repeaters, bridges, and routers to extend, as well as segment, local area networks.

A *repeater*, as in a cable TV network, connects two segments of network cable. It retimes, regenerates, and forwards a digital signal. Repeaters, however, can only extend a high-speed LAN a few thousand yards.

*Bridges* are used to connect two networks which use the same network signaling and the same media access-control protocol, such as Ethernet.

*Routers* are used to connect two different types of networks, in this case to route IP datagrams.

## Internet Backbone

Routers are also used to connect LANs to WANs, and make up the Internet *backbone*. A backbone is a central network to which other networks are connected.

It is important to remember that in the same way that a Local Area Network is a network of computers, the Internet is essentially a network of networks, consisting of thousands of computer networks interconnected by routers. It is also important to make a distinction between an Internet backbone router and a router on the community Ethernet network, or on the customer LAN, as illustrated in later scenarios.

## Sample Cable TV Networking Protocol

An example of a protocol in use today for extending a LAN by cable TV is *UniLINK*™, developed by LANcity Corporation, which:

- provides two-way, symmetrical data transmission at a signaling rate of 10 Mbps
- extends Ethernet beyond its distance limitation of 2.2 miles out to 160 miles
- coexists with entertainment and other services on a commercial cable TV network
- provides the same data rates and services that an Ethernet provides.

The cable data modem is specifically designed for cable TV networks. It uses a bi-directional single or dual cable plant to provide symmetrical data transmission at a signaling rate of 10 Mbps. It uses *Quadrature Phase Shift Keying (QPSK)* and has a spectral efficiency of 1.67 bits/Hz. The data modem is frequency agile, allowing it to operate in any available standard 6 MHz channel, over a transmit frequency range of 10 MHz to 174 MHz and a receive frequency of 54 MHz to 550 MHz, with a bit error rate (BER) of 25 dB C/N: <1 in $10^9$.

## Hosts, Nodes, Gateways, and Routers

Internet backbone routers were originally called *gateways*. Gateways were developed to deal with the fact that for internetworking to work, computers communicating across multiple types of networks needed a way to talk with each other as well as to talk to the intermediary networks in between in order to pass packets. What was needed was an *end-to-end protocol.*

An assumption was made in protocol design that the networks themselves could not be modified in order to internetwork them. Therefore the gateways had to handle such things as differing maximum packet lengths and error characteristics among networks. The gateway would know about the end-to-end protocol used by *hosts* (end user computers) communicating across multiple networks.

Other terms that may be encountered include *nodes*, or *packet switch nodes (PSNs)*, which were also originally called *interface message processors (IMPs)*. These terms all refer to packet switches in the Internet backbone. An important distinction is that the term node when talking about the Internet means something entirely different than it does when talking about an Ethernet. A node in Internet terminology means a router; a node in Ethernet terminology is the end user computer, or the Internet's equivalent of a host.

## Cable Internet Brouters[iv]

A cable Internet brouter is a combination bridge/router designed to work over commercial cable TV channels, and has a form-factor similar to a standard set-top box.

Using a cable Internet brouter, all packets that are destined for users on the Metropolitan Area Network (MAN), or the community network, are transparently bridged using the IEEE 802.1D Spanning Tree protocol. In other words, packets destined for the community network are handled by the bridge portion of the Brouter.

Routing tables, which determine where to send packets that are destined outside the community network, or somewhere out on the Internet, are maintained using a routing protocol called RIP1. RIP1 is an *interior gateway* protocol used to execute distributed routing and reachability algorithms with other routers.

For packets going out to destinations across the Internet, or IP packets, the cable Internet brouter conforms to specific Internet protocols including the Internet protocol (IP), Internet Control Message Protocol (ICMP) and others when required. This ensures that other computers in the network receiving the data packets will be able to process and route them to their destination, i.e. that they "speak the same language".

The brouter interfaces between an Ethernet LAN and the cable TV network and performs required functions such as encapsulation and decapsulation of the data (i.e. putting the data in an envelope with an address on it, and taking it out of the envelope), sending and receiving datagrams, performing IP destination address translation, and network flow control and error handling. All of this takes place transparently to the user, but internally a complex system of data transmission, traffic control, and error handling ensures that data reaches its destination intact, and in a timely manner.

The brouter receives and forwards Internet datagrams, providing buffer management, congestion control, and fairness. This activity allows data waiting to be sent to be stored in holding areas, or buffers, and ensures that each user "gets a turn" to transmit or receive data when the network is being heavily used, or is "congested", the equivalent of a system of onramps and traffic lights to control traffic during rush hour.

The brouter also chooses a *next hop* destination for each IP datagram. This means that the network chooses which computer to send the data to next. Each computer in the chain from sender to receiver is called a *hop.* Internally, the network keeps track of the best routes for data at any given time, dependent on various conditions in the network. For example, it may have detected that a certain computer is not operational. Even though taking this route may be the fewest "hops", and hence the natural first choice to send the data to, it will choose an alternative route.

For non-IP packets, the Internet brouter uses the Spanning Tree algorithm and automatically learns the locations of devices by listening to network traffic, forwarding packets only when necessary. Packets that are not forwarded are *filtered.*

Realizing that a standard communication protocol is needed for cable TV based broadband communications networking, the IEEE 802.14 working group was formed.[v]

The 802.14 protocol will define multiple physical layer protocols, a MAC layer protocol, cable topologies supported, and other criteria.

## CABLE DATA NETWORK MANAGEMENT

Largely due to the decentralized nature of the Internet, there is no central network management. Each IP network is responsible for managing itself. Out of necessity, evolved a common technology used to manage the individual components of the Internet, which, like many other protocols originated for the Internet became adopted as the standard for non-Internet networks as well.

This management framework is called *Simple Network Management Protocol (SNMP).* SNMP evolved from the *Simple Gateway Monitoring Protocol (SGMP)* which was designed for monitoring IP gateways in wide area networks.

SNMP is the defacto standard for network management today. By using SNMP network management, cable Internet brouters and bridges are fully integrated with the network management of all of the other IP networks on the Internet.

SNMP allows the collection of network statistics from widely diverse network components by defining the minimum amount of information that each IP device should provide, via a structure called the *MIB (Management Information Base).* An SNMP *agent* is the software which interfaces between the MIB and *the network management station (NMS)* and processes all of the MIB and management requests and responses to a device. It uses the SNMP protocol to package the request and responses.

In a cable bridge or Internet brouter, the implementation of an SNMP agent supports MIB II objects (an extended MIB to cover different types of network devices) and includes proprietary extensions to the MIB for management of cable TV objects. Also included are the UDP and IP protocols required to exchange SNMP packets with a network management station.[vi]

It should be noted that these network management capabilities offer cable operators an added benefit by providing additional information for troubleshooting the physical cable plant and thus improving reliability of the entertainment network as well.

## CABLE DATA NETWORK SECURITY

Once one computer is connected to another, some security risk exists, no matter how slight. It is often said that the most secure network is no network at all. The more connections there are, the greater the risk, and the greater the potential that the network can be broken into by a *hacker.*

Security can have many different definitions, but basically includes the areas of data integrity, user authentication, and privacy.

According to Al Hoover, vice president of information and application services at ANS, there are three questions you want to ask[vii]:

- What are you protecting?
- Why are you protecting it?
- What are you protecting it from?

The most common security risk in a computer network is also the easiest to overlook: password security.

The first level of security planning is also the toughest: ensuring that users do not pick passwords that are easy to guess, such as the names of family members. Password generation programs are sometimes used to inhibit this. Hackers will sometimes use stolen password lists in what is referred to as a "dictionary attack". Another possibility is that the hacker has a password capture program, whereby when a legitimate user logs in, his or her password is "captured" and reused by the hacker. Training users on password security is a key factor in creating a secure network.

For a community-wide Ethernet network connected to the Internet, security is addressed on several different levels.

## Closed User Groups

From the community side of the network, security can be provided via a system of *closed user groups*, as implemented in the UniLINK protocol. Using this system, users are assigned to one or more user groups. Ethernet data from one bridge or brouter can only be read by another bridge or brouter if it is a member of the same user group. This system allows multiple users to share the same RF channel, but operate as if they are on different networks, essentially creating multiple *logical* networks on one *physical* network. Schools could be on one logical network; hospitals, medical centers, and doctors' homes on another, and a business and its telecommuting employees on a third.

## Filtering, Authentication, and Scrambling

Security is also provided in a community Ethernet network via features such as IP address filtering, traffic type filtering, broadcast and route filtering. In addition, built-in security features, including user authentication and passwords are provided to prevent tampering with the cable TV receive and transmit frequencies.[viii] The fact that the data is modulated using QPSK and the data is scrambled using changing seed patterns means that it would be extremely difficult to decode.

## Firewalls

One security device often used in corporate networks is a *firewall.* A firewall creates a safety mechanism whereby all traffic on the network must pass through the firewall computer, and only authorized traffic is permitted to pass. A firewall can be between two private networks, or between a private and a public network, such as the Internet.

## Application-Layer Gateway

Another security option is an application-layer gateway. With an application-layer gateway, the user is not directly connected to the Internet, although to the user it appears as if he is. The gateway actually executes the user commands. The gateway can also examine the source and destination address and commands entered. A user accessing the Internet must pass an authentication process to gain access through the gateway. The gateway can also be set up to prohibit a user from using certain commands which could pose a security risk to the organization.

## Tunneling

One risk to firewall security is the practice of *tunneling.* Tunneling can have positive uses, or can be used to circumvent security. Tunneling refers to the practice of encapsulating a message from one protocol in another, and using the facilities of the second protocol to traverse some number of network hops. At the destination point, the encapsulation is stripped off, and the original message is reinjected into the network. In a sense, the packet burrows under the intervening network nodes, and never actually sees them. There are many uses for such a facility, such as encrypting links and supporting mobile hosts.[ix]

## REGULATORY ISSUES

The legal and regulatory environment in the cable and telco world, coupled with the growth of data networks like the Internet, has created the need for major reform in telecommunications policy in the US, Canada, and countries around the world. The regulatory environment is politically complex, and the need to relax existing cable and telco regulations in order to build a global information infrastructure is recognized.

The Internet, on the other hand, is currently unregulated. This is in keeping with the fact that common carriers are considered natural monopolies, and therefore are regulated, but services provided over common carriers are not. This regulatory policy is not consistently administered, however, since services provided over telco lines are regulated. This has caused some public interest groups to call for similar regulatory requirements for the Internet.

The issue of universal access is also integrally tied to the debate over regulation of the Internet. The concern is that the Internet not create a society of "haves" and "have nots" on the global information infrastructure. The current thinking is that a certain level of "basic service" should be available to all citizens for a modest fee. But the debate rages on as to what defines "basic service" and how to subsidize such universal access.[x]

Given the high "public good" quotient that such networks bring to municipalities, schools, and the general citizenry, in addition to businesses and for-profit enterprises, it is unlikely that government regulatory agencies will regress into imposing a stricted-regulated environment that would hinder the growth of the national information infrastructure.

## COMPARISON OF INTERNET ACCESS OPTIONS

The term "Internet access" is often used as if there is only one type of access available. In fact, Internet access runs the gamut from low speed dialup access to high speed dedicated network access. Below are some of the primary alternatives.[xi]

### Dedicated Access

Dedicated access is appropriate for institutions and businesses who want to be hooked up to the Internet. For dedicated access, the user needs:

- a dedicated leased line (56 kbps or faster)
- a router

This kind of connection in the US costs at least $2,000 initially, with monthly fees starting at $1,500 per month to much higher charges as line speed increases. Dedicated access allows all computers, and all users on a Local Area Network (LAN) to connect to the Internet through the router. It allows users access to the full functionality of the Internet. Because of the cost of a dedicated connection, this is not a practical option for home users.

### SLIP and PPP

A less expensive option is to use standard phone lines and high-speed modems, and connect to the Internet using the Serial Line Inter-net Protocol (SLIP) or Point to Point Protocol (PPP). For dial-up access, the user needs:

- SLIP or PPP software
- a high-speed modem (V.32 or higher)

With SLIP and PPP, the user has full access to Internet resources, but saves on the high connection costs of a dedicated leased line. With these options, the user is actually "on the Net", as opposed to accessing it through another system. They are suited to connecting an individual or home user to a larger LAN, which has dedicated access to the Internet. This solution is not appropriate to connect a network of any size to the Internet, due to the line speeds.

The connection to the Internet is provided by a national or regional access provider. A provider like PSI or UUNET charges about $250 per month for unlimited SLIP or PPP service; alternately, there may be a lower monthly charge, with an additional hourly fee. Service providers may supply 800 numbers or local access numbers in major urban areas to minimize telephone costs.

**Dial-Up Access**

The low-end option for Internet access is a simple dial-up connection to a computer which has dedicated access. For dial-up access, the user needs:

- a terminal emulation package
- a modem

In this case, the user is not really "on the Net". The advantages are the it is cheap and easy to set up. The disadvantages are that the user may not be able to access all Internet services, and is dependent on the service provider for services and disk space. The user may also have to pay the phone charges, if 800 numbers or local access numbers are not provided.

The cost of service is typically $20 to $40 per month, (possibly with some additional per-hour access fee). The cheapest rates apply if you contract for "off peak" service only (i.e. nights and weekends.

**Cable TV Internet Access**

The advantage to the user of accessing the Internet via cable TV is that he does not need his own leased line in order to get high-speed 10 Mbps access to the Internet. In the past, these speeds, since they required a dedicated leased line, were out of reach for all but large corporations. By connecting businesses, institutions, municipal offices, and home users into a community wide network over cable TV, performance can be improved by three orders of magnitude while cost is reduced.

## COMPARATIVE INTERNET ACCESS SCENARIOS

This section depicts several typical Internet access scenarios and compares and contrasts them with Internet access via cable TV.

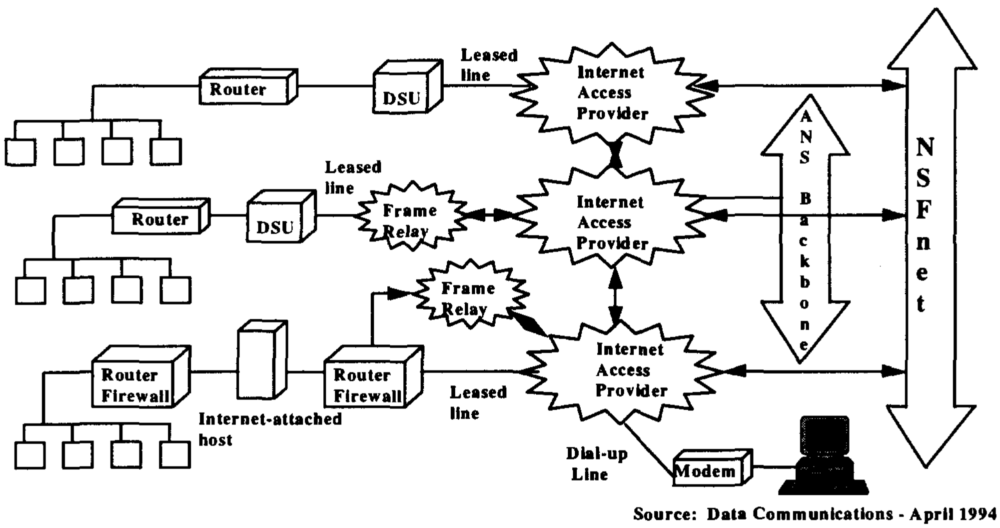


**Figure 1 - Corporate Connectivity to the Internet**

## Telco Models

One typical method for corporate connectivity uses a router and *Data Service Unit (DSU)* to tie directly into an the Internet Service Provider's network. A methodology called *frame relay* can also be used for this purpose. Figure 1 shows a typical scenario for connecting corporate users to the Internet. In this scenario, the remote or home users can log on via modem and dial-up line[xii]

Typically, corporate users access the Internet via leased lines. The company buys one or more leased circuits that connect one or more routers at the customer premise to a router at the Internet service provider's site. Available line speeds range from 56 Kbps to 45 Mbps (T3). The Internet service provider may supply the router at the customer site.[xiii]

Pricing for such services vary. One Internet service provider, Uunet's Alternet, has pricing which begins at $795 per month for a local 56 kb link, T1 starts at $1,250 per month.[xiv]

One key point is that the customer, and not the Internet service provider, is responsible for procuring the leased lines.

Another option that may be used by small companies, who do not need the high bandwidth that leased lines provide, is frame relay. PSI and Sprint, for example, offer frame relay services. In this scenario, users need a router with a frame relay interface and leased line to the Internet Service Provider's point of presence (POP). In this case, unlike leased lines, customers only pay for the bandwidth they actually use.

PSI's Interframe includes a router and frame relay DSU installed at the customer site. Committed information rates (CIRs) - the maximum average speed of the connection - range from 56 Kb to T1. PSI's prices start at $400 a month and extend to $3,400, depending on management and equipment. Sprint offers a zero-CIR service (no maximum average speed) that allows for bursts of traffic at up to T1 for a flat rate of $400 per month.[xv]

## The Cable TV Models

There are multiple alternatives for the configuration of Cable TV networks which offer Internet routing.

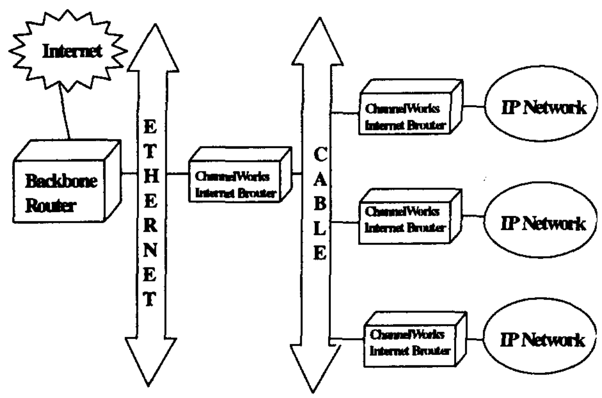


**Figure 2 - Multiple IP Networks on a Cable Network**

Figure 2 shows multiple IP networks on a cable network connecting to the Internet using Internet brouters. In this scenario, there are multiple IP Networks on each cable network. This is a practical scenario for a community which includes multiple IP addresses which requires IP routing with cable TV connectivity. In this scenario, each site could have its own IP network, and each would have an Internet Brouter. Each site would have its own IP address space, and its own IP administration. The security firewall would be at the brouter at each site. This scenario would allow for increased security, ease of management and administration, expansion (the ability to add more sites/IP networks), plus freedom and ease of communication.

A second scenario is having a single IP network on each cable network, as depicted in Figure 3. In this configuration, the entire cable TV network using the bridges will be a single IP network. In this scenario, all sites would share the same address space, and would have common IP administration. The security firewall would be at the single router entry point, and there would be no IP firewalls between sites.

This implementation might not be practical for a city with multiple businesses and institutions using the same network, and would require coordinated management of IP addressing for all of the users.

Ultimately, in a third scenario, multiple cable TV networks could be connected together as an IP network over a WAN, via a long distance carrier such as MCI or AT&T, or a Competitive Access Provider (CAP) such as Teleport. In this scenario, the Cable TV networks could either be bridged or routed.

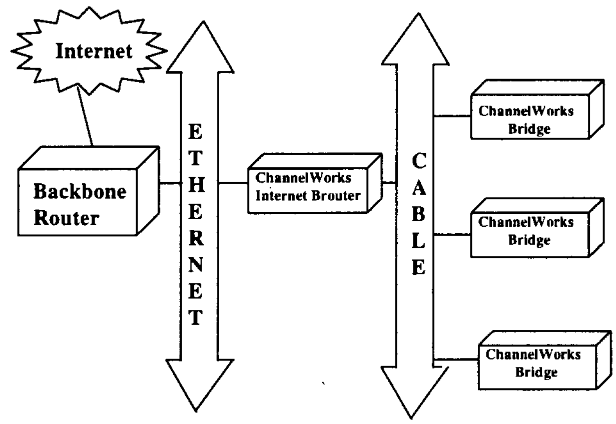


**Figure 3 - Single IP Network on Each Cable Network**

## CASE STUDIES

### Electronic Commerce Network ECnet - Phoenix, Arizona

ECnet was one of the first data networks set up over cable television. The network was developed as a collaboration between Times Mirror Cable Television, Digital Equipment Corporation, and Arizona State University. ECnet connects together manufacturing companies in the Phoenix area for the purposes of concurrent *Computer Aided Design (CAD)*, video conferencing, *electronic whiteboarding*, and access to the Internet. Companies using the network include McDonnell Douglas, Tempe Precision Aircraft, and Modern Instruments.

The backbone of the ECnet network is a 100 Mbps *FDDI* (Fiber Distributed Data Interface) fiber ring, which connects four headends. Connected to each headend is a community Ethernet network, comprised of one to three companies. Each company site itself houses its own LAN, which may be comprised of hundreds of users. Bridges are used to connect the LAN sites to the community network.

The physical media in the network include dedicated fiber in the backbone, shared AM fiber for the headend trunks, and coax. The fiber ring is supporting distances as great as 36 miles between headends, while the longest fiber/coax headend trunk extends over 15 miles to a customer site. The network operates downstream at 336-342 MHz, and upstream at channel T8 (11.75 MHz - 17.75 MHz).[xvi]

Access to the Internet is provided by Arizona State University. Security is provided which includes 24 hour/7 day monitoring, file encryption, protocol monitoring, automated alerts, and lockouts.[xvii]

### Hawaii Public Schools

The Hawaii public school district includes 360 schools located on six islands. This unique geography has perhaps contributed to the Hawaii Department of Education being on the leading edge of networking technology.

Oceanic Cable, a Time-Warner subsidiary, Digital Equipment Corporation, and Convergence Systems, Inc., a Digital reseller, have collaborated on the Hawaiian school network.

The Hawaiian schools original network consisted of a T3 (45 Mbps) microwave backbone, 28 T1 (1.5 Mbps) leased lines, which didn't meet the needs of the school system. "Leased lines are expensive and they don't really provide us with the bandwidth we need for the applications we'd like to run on the network. We needed a high-bandwidth, high-speed network. With the telephone company, that would have meant a T1 line for every school, which would have been very expensive", said Kyunghak J. Kim, director of network support services for the state of Hawaii's department of education.[xviii]

Hawaiian schools are using the Internet for collaborative learning with schools on the mainland, accessing images from weather services, maps, and information from libraries and universities. "The main thing ChannelWorks™ has provided students is the ability to effectively communicate with other students in other parts of the country and the world, and the capability to access resources available in other places," comments Kim. The ChannelWorks solution has sped up Internet access considerably at Hawaii's schools: sending a message from the University of Hawaii to the mainland and back can now be completed in seven or eight seconds - which is 100 times faster than what was possible on phone lines.[xix]

## HOW TO SET UP A COMMUNITY ETHERNET TO INTERNET NETWORK

Setting up a cable television network for Internet access involves both business and technical issues. To begin, a business case must be completed, and the scope, schedule, and budget for the network must be defined. A project team, project manager, installation team, and site contact for each site must be identified, and a project plan prepared.

### Network Planning and Design

The next step is the network design and network map. The design must include network layout, site locations, amplifiers, channel assignments, network components, leased lines, etc.

The basic requirements to set up a community Ethernet to Internet network are:

- one forward and one reverse channel
- one bridge or brouter per site
- one translator at the headend (not required for a dual system)
- diplexors (either sub-, mid-, or high-split)
- an Internet point-of-presence (i.e. access to a backbone router on the Internet) either via one of the user sites on the network, or via the headend

Several decisions need to be made during the network planning phase, including:

1. Which sites will be in the same Closed User Groups?
2. Which sites will be on the same IP network?
3. What are the security and firewall requirements, and where should firewalls be located?

4. Where will the connection to the Internet be located? Will it be in the headend or at a user site?
5. If the Internet connection will be at the headend, who will install and manage the connection?
6. Who will procure, assign, and manage the IP addresses?
7. Who will secure the leased line connection to the Internet Service Provider? What line speed is required? How much will it cost? How will it be paid for?
8. Who will manage the data network?
9. Who will supply help desk support? What is the problem reporting procedure? What are the service hours? What are the problem escalation procedures?
10. What is the monthly service charge? What are the billing procedures?

The services of an Internet consultant or Internet Service Provider, particularly one who is familiar with cable television, may be helpful during the network planning and installation phase of the project to help answer these questions and devise the network plan.

As early as possible in the project planning, a certification of the cable plant should be done. The certification of the cable plant is an important first step. A detailed checklist is used to ensure that all requirements are met so that the network will function properly.[xx]

The first requirement for the network is that one forward and one return channel must be allocated for the network. The transmit frequency range is 10 to 174 MHz. The receive frequency range is 54 to 550 MHz. The plant must be two-way, with two-way amplifiers installed and activated, drops must be installed to all sites, and the channels to be used must have no ingress, extraneous carriers, or other signals in either channel. At this phase a 24-hour sweep on both channels is recommended to make sure there is no time-related ingress on either channel.

**Network Installation**

When the above steps are completed, the sites are ready for the installation of the networking equipment. At this phase, a backbone router and a leased line are installed at the headend, if required.

**Network Management**

Network management may be supplied by the cable operator or a third party. As a network grows in size, so does the need for network management. The plans for network management should be put in place during the planning and design phase.

## CONCLUSION

The capacity of the Internet will continue to grow, as will the user systems connecting to it. Multimedia and client-server based applications, such as digital libraries, telecollaboration, concurrent engineering, and visualization will proliferate. Multicasting, taking video or audio material, digitizing it, and sending it over the Internet, will benefit greatly from cable. Videoconferencing over the Internet to the home PC or to the desktop will become possible. Cable TV can make high speed access to the multimedia Internet of tommorrow a reality today.

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# MANAGEMENT ASPECTS OF A HYBRID FIBER COAXIAL (HFC) NETWORK

by
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*Abstract*

*This paper addresses Network Management aspects of a Hybrid Fiber Coaxial Full Service Network (FSN)* [i] *delivering services such as video, voice, and data. Spectrum Allocation and Bandwidth Management structured around an Open System Architecture are some of the focal points of this paper.*

*Illustrated in the paper is a design approach describing an example of the Network Management Architecture. It is based on an Open System Architecture for the new management platform because it promotes connectivity to existing Management Systems (Operation Support Systems). In addition, this platform makes it possible to incorporate a more intuitive, standard Graphical User Interface (GUI) in a windowing motif.*

*Also addressed is the selection of off-the-shelf solution tools for presentation, communication and the storage of the distributed information.*

*Use of Artificial Intelligence (AI) techniques in decision-making tasks for a distributed network environment is suggested as a logical extension to the HFC network management.*

## HFC Network Overview

CATV networks of today are evolving into two-way switching and distribution systems carrying both analog and digital services that are now referred to as FSN. As fiber gets closer to the home (Figure 1), networks are able to offer more services (such as video, voice, high and low speed data, and telemetry). As a result,

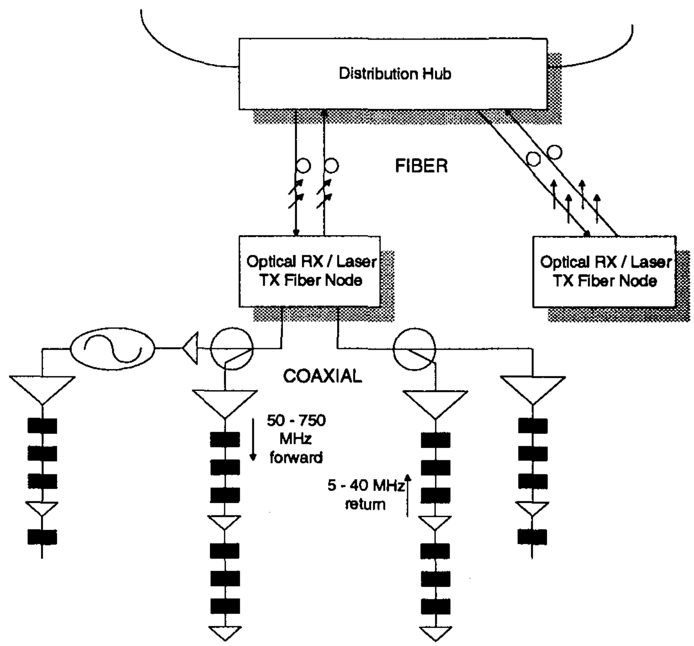


**Figure 1 Fiber to the Node HFC Architecture**

today's Network Management Systems (NMS) must provide management features never before required in the older one-way video broadcast system.

Managing HFC networks is complicated by the fact that these networks asymmetrical and were built without attention to many management aspects. Further use of equipment from various vendors with proprietary protocols increases the complexity in the implementation of a standards based system. The complexity of mapping many different services to the HFC's asymmetrical bandwidth dictates the use of a centralized Spectrum Management entity.

## Network Management Paradigm Overview

A NMS consists of three main elements: a manager, agents, and managed objects. The manager is the console through which the network administrator performs network management functions. Agents are the entities that interface with the actual devices being managed. Optical hubs, trunk amplifiers, and Premise Interface Devices (PIDs) are examples of managed devices that contain managed objects. Examples of managed objects might be Automatic Gain Control (AGC) signal, provisioning information for the PID, or anything else that directly relates with the particular device.

The Managed Objects are described in a structured way using Abstract Syntax Notation 1 (ASN.1). As the name implies, ASN.1 is the formal notation used to describe managed information in a formal and unambiguous way. All managed objects are structured using a framework called the Structure of Managed Information (SMI). A Management Information Base (MIB) defines managed objects and is written using ASN.1.

There are several protocols used for network management. Among the most popular ones is Simple Network Management Protocol (SNMP) which uses ASN.1 as a tool to describe its MIB. SNMP is a "light weight" protocol and imposes minimal overhead for implementation. It is an accepted standard in the Internet community and appears to be a preferred protocol for HFC network management by the industry. For interoperability requirements, vendors with proprietary protocols can encode data using Bit Encoding Rule (BER) which is available under SNMP.

To accommodate devices that cannot contain a full standard compliant agent, the concept of "proxy" is used. In this scheme, a standard compliant agent, typically a software entity residing on a PC in the Network, acts as a proxy for one or more other devices. This is commonly used whenever there is a need to interface a proprietary protocol with a standard protocol.

## Spectrum Allocation And Bandwidth Management

The main focus of bandwidth management is to address the bottlenecks in the coaxial plant, specifically in the return direction where the bandwidth is limited. Network management must pay close attention to this area so as to distribute two-way services in an efficient manner. Overall though, the bandwidth management encompasses the entire HFC from the Distribution Hub/Fiber Node and the rest of the coaxial cable to the Subscriber Premises.

The basic spectrum allocation operation would fit in Network Management in the following way:

The entire spectrum would be managed from the Spectrum Management Application (SMA) which would reside in the primary Headend or Distribution Hub as a part of the NMS.

On request from the SMA, the spectrum allocation operation would be carried out by the Spectrum Allocation Proxy Agent (SAPA) which would reside in the Distribution Hub.

Between the Distribution Hub/Fiber Node and the rest of the coaxial network, it is typical to have a proprietary transport protocol in order to carry out spectrum allocation. That would involve using a separate, maybe out-of-band, management channel.

The whole decision making intelligence that determines spectrum allocation to different services would reside in the SMA.

To be able to address spectrum allocation and bandwidth management, it is necessary to identify all Spectrum Management Objects (SMOb) and to describe them in a rigorous way in order to avoid misinterpretation.

From the Spectrum Management point of view, the HFC can be viewed as several abstract networks layered over each other. Due to the predominantly asymmetrical architecture of the HFC network and the diversity of services on it, the abstract layers may be unidirectional. The SMObs are implemented on the physical devices. They can be categorized in several groups: Abstract Network Layers Objects, Service Objects, Channel Objects, Modulation Objects, Power Level Objects and so on. These objects are interrelated in a structured and behavioral way. The best way would be to describe the system by describing the MIB using ASN.1. However, that would go beyond the scope of this paper.

From the implementation point of view, the managed objects would be distributed over the network and the communication /control of them would be done through a combination of standard and proprietary protocols (Figure 2).

The ultimate goal of spectrum management is to simply provision a service and the management application and dynamically allocate spectrum for the service through automatic provisioning of the network devices involved. This can be accomplished through autodiscovery and enough intelligence for the software to reprovision devices for the most efficient use of all resources. Then together with the fault management application, both will monitor and reprovision the system to meet the set requirements of operation (Figure 2).

## Off-the-shelf Solution Tools

Our assumption is that the HFC will be built based on cost/performance criteria. That implies using distribution/switching equipment from various vendors. We want to consider the development tools that will meet the accepted standards for interfaces in this system.

The network management development tools can be subdivided in several groups: Core Management Platform, Communication, and Database. We will address each group separately.

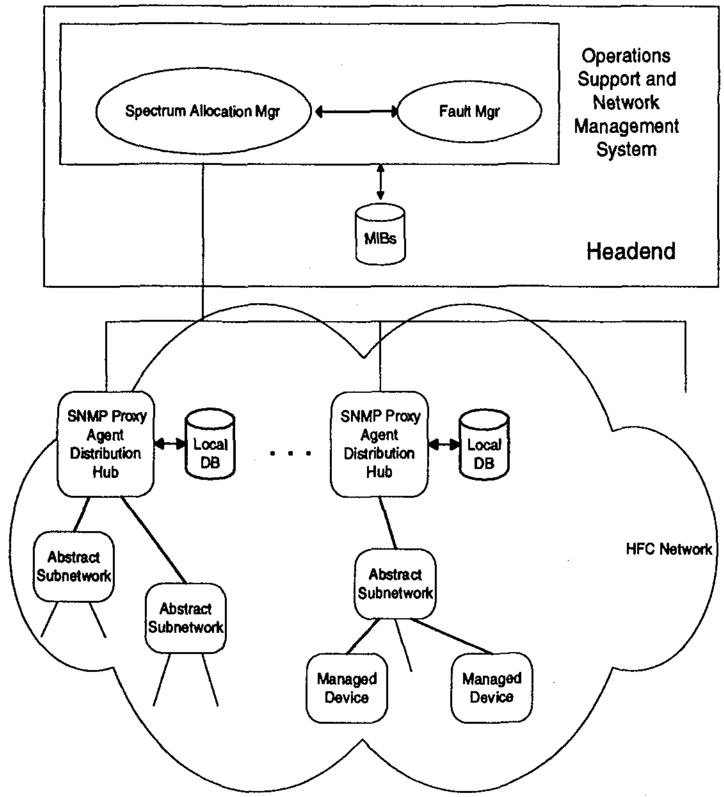


**Figure 2 Intelligent HFC Network Management**

Core Management Platform (CMP)

In general, Core Management Platform provides an integrated environment for displaying information from several different applications at the same time. It does this by combining device and status information from each application. All applications run in the background. CMP controls exchanging of the information between the platform and the applications and updates the graphical user interface accordingly. It usually provides the hierarchical graphical map with interconnected device icons and the mechanism to create and edit the map. Among other things, it provides a way to handle the alarms as well as "autodiscovery" and "layout" of the network. In this context, autodiscovery is a mode of operation where the management system "discovers" the managed devices and draws the icon based map of them in the layout mode.

Several major companies provide the CMPs. The key element in Core Management Platform is scalability. It is important to select a platform that can be applied to smaller networks but leave the possibility to grow to virtually limitless size. Very often you may need to manage a small network using a PC based development platform, but you want to integrate it with some other more powerful workstation based platforms. One such CMP is HP OpenView (HPOV). It is based on an international OSI management framework and models. In particular, it derives three essential elements from the OSI standard: a network management framework, a well-defined mechanism for describing managed objects, and a set of services and protocol for communication. At the same time it supports the SNMP concepts.

Communication

Exchanging the management information between the Managed Objects and the managing station requires some information exchange infrastructure. This infrastructure is usually different from the infrastructure that the network uses to communicate the payload information like voice, data, and video. For example, in HFC networks it may be convenient to use an out-of-band channel to provide the communication for management information.

As mentioned earlier, there are several standards for managing networks being used in industry. Besides the syntax to describe the MIB, the standard defines the transport protocol to communicate information. This paper stresses SNMP as the most feasible choice. SNMP defines how to communicate the information between the management entity and the managed object. One of the transport protocols used in SNMP is User Datagram Protocol/Internet Protocol (UDP/IP).

There are some other proprietary transport protocols such as Sequenced Packet Exchange/Inter-network Pack Exchange (SPX/IPX) used by Novell. CMPs usually support both, although UDP/IP is the more dominant.

CMPs such as HPOV support SNMP and SNMP implementation tools from other sources. Some implementation tools comply with additional standards, and we will mention some of them used in the PC environment. For example, the WinSNMP standard defines a set of Application Program Interface (API) links to provide SNMP services. Also, WinSock provides a similar set of interfaces to perform TCP/IP communication.

Database

From the point of the Database Management Tools, a HFC network can be perceived as a Distributed Database. We need to manage a set of objects that are grouped in various network elements and distributed in a geographical area. The CMP offers some database capabilities. These are usually some Internet Protocol (IP) queries and some Alarm Database capabilities such as reporting of acknowledged alarms. The general idea is to build on these capabilities. As far as the database tools are concerned, three alternatives available are: 1) use relational database technology, 2) use Object Oriented technology, and 3) use a combination of the two. The network management as described by various Bellcore standards using Transaction Language 1 (TL 1) does not support object oriented technology. Most of the standards, including SNMP, are at least somewhat based on object oriented technology as it relates the protocol to the database.

There are two other important points. The database development tool would need to support Object Data Base Connectivity (ODBC). That is necessary to interface with the database modules provided on the CMP. The other aspect is to be able to let your product be flexible and independent from the particular vendor. It is important to select database development tools that support some type of GUI; this provides an integrated solution in representing physical entities as database elements. There is more than one database development tool on the market that supports all these requirements.

## An Example of Network Management Architecture

In our example, the devices in the network either will be SNMP compliant (contain an SNMP agent) or will be proxied by some other entity. Encompassing these two approaches in the Spectrum Management architecture will allow for a low cost solution, vendor independence, and compatibility with legacy devices upon which the newer HFC is evolving.

The Proxy Agent (Figure 3) is a Vendor Supplied Application which uses an SNMP interface between itself and the NMS. This approach can be implemented on a single controller platform (PC with MS Windows®™) that realizes an inter-application message passing operation through SNMP. The main function of the Proxy Agent will then be to translate the SNMP messages into a Vendor Dependent Protocol (VDP). The physical interface for the Proxy Agent on the vendor device side may be RS485, RS232, 802.3. which will connect to a RF Mux/Mod device that facilitates the communication with non SNMP managed objects/devices. This gives the industry a low cost solution of integrating the existing equipment that is not SNMP compliant into a single integrated open NMS. Typically, these RF Mux/Mod devices are non SNMP managed objects themselves and communicate over the HFC through a low data rate out-of-band channel.

The second approach is to include an SNMP agent into the new managed objects/devices of the evolving networks. And incorporating this functionality into the devices allows the cable

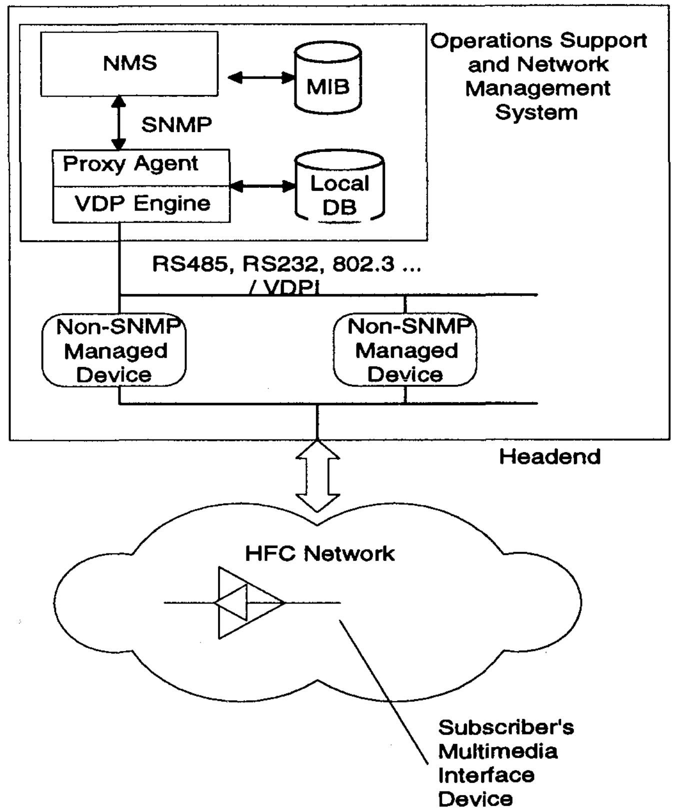


**Figure 3 SNMP Proxy Agent**

operator to build a vendor independent system. With this approach, the NMS can physically connect to the SNMP managed objects (typically the service specific Mux/Mod) at the headend via an 802.3 interface (Figure 4). However, for the NMS to manage the remote managed objects (SNMP or non-SNMP compliant) it must use a hierarchical approach. This is accomplished by first communicating with the service Mux/Mods, and then the service Mux/Mods will communicate with the remote managed objects units via the HFC.

## Future Directions

So far we have addressed the current technology that is being deployed for management of the HFC networks.

Traditional algorithmic approaches for network management have been successful for managing the networks of current complexity. As the networks evolve in complexity, the more powerful heuristic approaches such as Artificial Intelligence (AI) may be necessary.

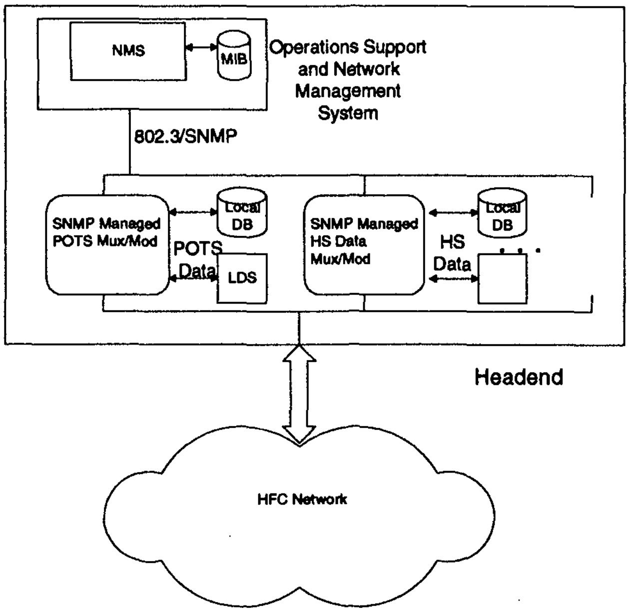


**Figure 4 SNMP Managed Devices**

AI is a very broad discipline and we will not go into it in any depth. Among other things, AI is a set of programming methodologies that focuses on the techniques used to solve problems by generating new strategies and plans. Expert systems are a subset of the AI disipline. The name is based on the term used in daily language for somebody who acquired experience that allowed him to solve problems. The programs that capture their problem solving strategies and selectively apply them under specific circumstances are called expert systems.

Many expert systems have been deployed in telecommunication industry in the past. Unfortunately, most of them were designed to solve local and constrained problems without addressing the needs of the distributed networks. That is changing, and today there are several laboratory expert systems that address the network management.

To illustrate how expert systems can be used to address the needs of HFC network management we will identify the functional tasks and corresponding modules to do them.

The modules/tasks can be categorized as follows (Figure 5):

1. HFC Distributed Network Model which represents the network configuration in terms of number of subscribers, traffic characteristics, managed devices and their attributes.

2. Optimal Resource Allocation Module which feeds its decisions to Knowledge base and Self-learning module.

3. Self-learning module which processes optimal solutions, makes proper changes in HFC network model and communicates it to the Knowledge database.

4. Knowledge Database stores the clauses that characterize the normal and abnormal operations of the HFC network.

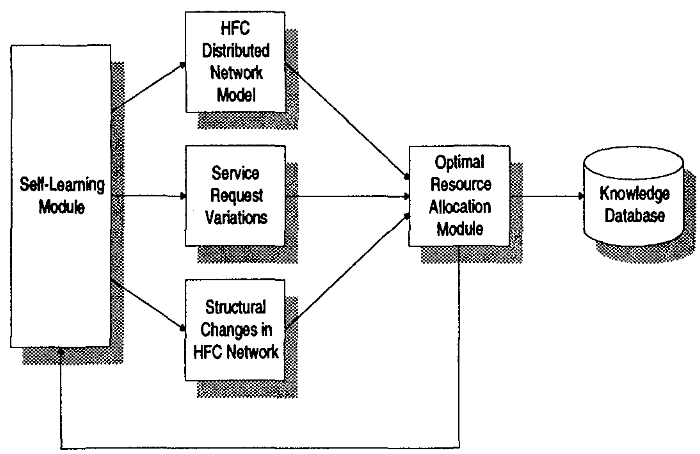


**Figure 5 HFC AI Self-Learning System**

Although the AI and expert systems of the future may be addressing needs that are hard to predict today, one application that is needed today may be the interoperabilty and communication among the equipment from different vendors. The expert systems would go through a learning process before they acquire knowledge of how to communicate with different pieces of equipment.

**Summary**

After reviewing basic HFC network architecture, we introduced general network management terminology in order to be able to address the needs of the HFC network management. Then we addressed the off-the-shelf tools and building blocks in such a way as to be able to develop a sound Standard compliant HFC Network Management System which would be able to manage the equipment from various vendors. Since the Spectrum Management of the HFC network represents a very important issue of the bandwidth utilization we addressed some basic aspects of it. At the end we illustrated our analysis by presenting a sample architecture of the Network Management System. The general philosophy emphasizes multivendors and standard compliance, while maintaining simplicity. Also, using off-the-shelf building blocks as much as possible improves speed-to-market of the product.

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# Managing the Return Spectrum to Optimize Interactive Revenue Opportunities

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Antec Corporation

## ABSTRACT

Issues related to reverse transmission in cable plant are considered. A topology is described which removes the largest portion of the interference in the return band. A few of the issues related to protocols suitable to return data are mentioned, and the idea of spectrum management is introduced.

## INTRODUCTION

For years the cable industry has had available technology that *almost* enabled the return band from 5 to 30 MHz. Some systems have employed the return band to transport video from local venues to the headend, and a very small number of systems have employed the return path to retrieve data from homes. However, little use has been made of the return spectrum until now. Cable operators have recognized that they will have to provide return for new services, however, and most systems now either have return path or can add it. In the days when systems were built according to tree and branch architectures (and there are a lot of such plants still in use), the return band was frequently considered impractical to use. However, with the modern fiber to the node architectures, the return band is usable.

A survey of several cable operators indicated expectation of revenue from services using reverse spectrum, ranging from $3.00 per month (average for 100% of homes served) to over $50.00 per month (in 25% of homes served). Revenue comes from such services as on line service (Prodigy, AOL, Internet, etc.), telephony (including long distance access) and interactive advertising.

## ISSUES IN SELECTION OF A FREQUENCY PLAN FOR UPSTREAM USE

Any attempt to set a frequency plan for upstream traffic must first take into account existing services if any. Most of the existing services are either video return from remote venues, or impulse pay per view systems with RF return. While we expect future services to be controlled with some sort of intelligent system, existing systems will continue to be controlled manually. Future control systems are discussed below.

Interfering carriers will also render some frequencies inoperable for all except perhaps very low data rate services. A review of the literature suggests that frequencies below 10 MHz are very difficult to use, and that frequencies above 20 MHz are much less susceptible to interference. A point of concern with the use of frequencies above 30 MHz, is the potential for interference to TVs and other devices connected to the cable. TVs in North America use the band from 41-47 MHz as their intermediate frequency. A typical level for a return signal leaving the home, is about 50 dBmV according to current thinking. This is as much as 50 dB above the incoming signal the TV is receiving. If a significant part of the upstream energy strikes the TV (due to limited coupler isolation), then interference is possible.

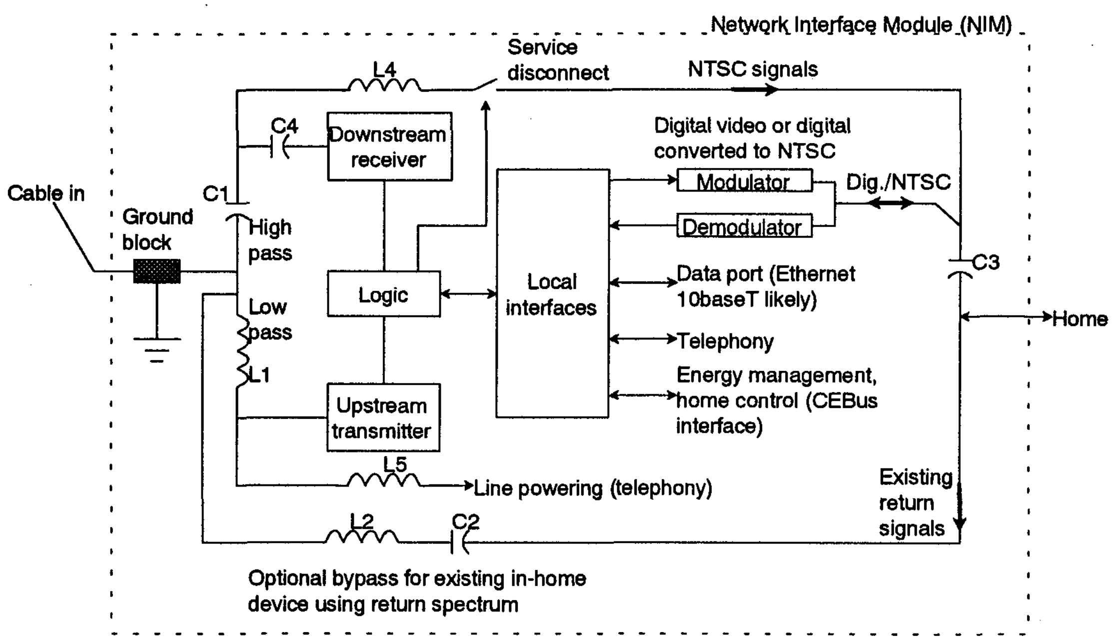


**Figure 1. Configuration of a Fairly Complex NIM**

## INTERFERENCE IN THE UPSTREAM BANDWIDTH

Many studies have been conducted of the interference in the upstream path. One of the few conclusions that may be drawn from these studies is that the interference varies considerably from one system to another, and within a system, from time to time. Most investigators do agree, however, that the primary source of interference is the home. Drop cable is responsible for the next largest contribution, but is much less of a contributor than is the home wiring. Finally, a little interference may be picked up on the distribution plant, but this is by far the smallest source of interference.

The above considerations lead one to favor an approach to return architecture, in which reverse signals are added in a box on the side of the home (preferably inside, to reduce the cost of meeting the temperature range), located ahead of any subscriber devices. In this case, it is possible to provide for a high pass filter on the cable entering the home. The filter will reduce interference generated in the home, and will also protect the subscriber's equipment from the effects of reverse signals in the 30-40 MHz band. Such interface boxes have been called Network Interface Modules (NIMs) or Network Interface Devices (NIDs). Other names we have seen include Coaxial Termination Unit and Premises Interface Device.

Figure 1 illustrates one possible NIM configuration. This is a rather complex NIM, in that it is provisioned to handle not only downstream NTSC television, but also upstream signals from an RF impulse pay per view (RF IPPV) box of the type in existence today, and bi-directional signals from a data port, a cable telephone and a home control system. The likely configuration for such a device is modular, so that only the functionality needed will be plugged in.

The signal enters through a ground block, which may be included if the NIM is mounted outside the premises, but not if it is mounted inside. The C1 high pass section couples all the signals above 50 MHz, including downstream NTSC and digital signals.[i] The L1 low pass section couples upstream signals below 40 MHz, to the cable. In a certain band, signals from an existing set top box having an RF impulse pay per view module, are coupled from inside the house to the incoming cable, through bandpass filter L2-C2. In this band, any noise coming from the home is coupled to the cable, rendering this spectrum no more useful than it is already.

The upstream transmitter in the NIM is able to utilize any return spectrum not used by the in-home RF IPPV device. Presumably any new upstream equipment is frequency agile and managed by a management system as shown below. Notice that, with the exception of the band passed by L2-C2, no energy in the upstream band is coupled from the home to the distribution plant, so the major source of return interference is eliminated. Only systems using RF IPPV will need to provide the L2-C2 bypass.

As shown, NTSC downstream and possibly digital signals, pass through the C1 high pass section, where they are presented to the inside wiring. A service disconnect is shown to allow for disconnection of all services. We are not sure if this will be demanded of a box that normally sits in a homeowner's basement or garage: the chance of bypassing the box may be so great that the service disconnect is not considered appropriate.

Low pass section L4 is needed to attenuate the digital (550-750 MHz) band under some circumstances. A downstream receiver accepts data signals that the NIM is to process. These signals may, depending on the services ordered by the subscriber, include telephony, data, home control, and perhaps even digital video programming. Elsewhere in the NCTA Technical Papers for 1995[ii] is shown a system in which digital TV signals are extracted in the NIM and either decompressed and converted to analog in the NIM, or transmitted into the house to a lower cost digital set top converter.

Additional services may be provided at this point. If data services are provided to the subscriber, many operators are favoring providing a 10baseT (Ethernet, twisted pair, 10 Mbps) interface. This interface is becoming standard on many personal computers, and low cost boards are available for computers not so equipped. The 10baseT interface uses RJ45 connectors (big brothers to the RJ11 telephone connectors), and twisted pair cable. The cable is much easier to wire than is coaxial cable. We question the philosophy of putting computer interface connectors on set top converters, because most people don't watch TV and use a computer in the same place, and wiring from the TV to the computer is likely to be deemed undesirable by many subscribers.

A second service contemplated by many cable operators, is the provision of cable telephony services. This may be provided from the NIM by inserting a line card, which carries all of the interfaces needed to support telephone services. The line card will make the telephone "think" it is connected to a central office switch, and the logic will handle the details of getting signals to and from the headend. Downstream signals are received by the downstream receiver. (In some cases it may prove more economical to provide a separate receiver for telephony service due to signal routing complexities.) The line card may be replaced with an ISDN card for provision of ISDN services, a class of service that has not penetrated the residential market to date. ISDN promises faster data services and simultaneous voice traffic with a single telephone line.

Of serious concern in the providing of telephone services, is the lifeline problem of powering the telephone when local power at the home is not available. Subscribers have long expected that their telephones will work when the power is out. Some operators are talking about not providing first line service because of the powering problem. Others are considering home powering with battery backup. This would normally work, but we must be concerned with battery reliability under temperature extremes, and that batteries do create maintenance issues in the long run. Still other operators are planning to provide enough network power to operate the telephone interface without commercial power. This power is either provided on a separate twisted pair "siamese" cable, or on the center conductor of the coax. Taps are available now for the siamese cable, and will be available shortly for center conductor powering. In the figure, L5 extracts power for the telephone equipment, from the center conductor of the drop.

Another service of interest to some operators, is energy management and home control. Some utility companies have indicated a willingness to participate in the cost of providing the NIMs in order to gain control of the homeowner's major appliances such as air conditioning. The air conditioning and other loads, are cycled on and off during peak demand times. Alternatively the power company may increase its rates during peak hours. This service gives rise to other possible low speed data services such as security monitoring. Such services are similar to the data services shown above, but differ in the data rate (very low) and the priority of communications: if energy management commands take a minute or so to be delivered, nothing is lost. Of course, this is not true for security services. The most likely interface for these lower speed data services is the power line communications standard of the EIA's Consumer Electronics bus (CEBus), codified in the EIA IS-60[iii] voluntary standard. Communications is by power line, so no special wiring is needed (though in some cases an RF coupler is needed between the two line phases).

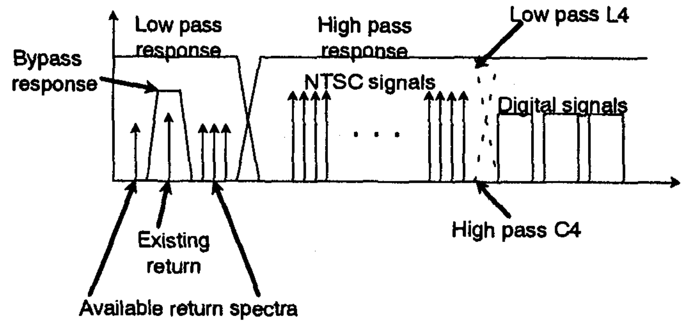


**Figure 2. Spectrum Management in the NIM**

Figure 2 shows the spectrum and how it is handled in the NIM. The 5-40 MHz reverse band is low pass filtered in L1 (figure 1), and contains all upstream signals. Most of the upstream signals likely will be generated in the upstream transmitter shown in figure 1. However, if an existing RF IPPV converter is in use, the bypass shown is provided by L2-C2. This spectrum bypassed would normally not be used for services other than RF IPPV.

The downstream spectrum is assumed to consist primarily of NTSC signals from 54 to 550 MHz, and digital signals above that. The NTSC signals are routed directly to the home as they are now, through L4, the service disconnect, and C3. The reason we show low pass filter L4 in figure 1, and in figure 2, is that some proposals involve reusing the 550-

750 spectrum in the home, for signals generated in the NIM. If this is done, it is desirable to attenuate the digital signals, which are not needed in the home for these scenarios.

## HIGH RETURN BAND

Some investigators have considered using a high return band around 900 MHz. This has reportedly been used in one or two systems. The likelihood is that in a few years, demand for return spectrum will exceed that available between 5 and 40 MHz, even in systems built with small nodes. We believe that use can be made of the high return spectrum, but that the demand is a few years into the future. Practical filtering topologies have been successfully tested in the laboratory.

## POSSIBLE DATA PROTOCOLS

Data services on cable are characterized by the almost universal need for two way communications. Some services such as telephony require symmetrical spectrum, with the same data requirements up and down stream. Others, such as most data services (on-line connectivity being a prime example), usually have need for more bandwidth downstream than upstream. Particularly in the upstream direction, the data flow is very bursty, with long periods of no communications, but with need for fairly fast response when a transmission is sent. Telephony requires low latency (that is, low delay), and arrival of packets in fixed time intervals (low jitter). Television signals require low jitter (or buffering, which can be expensive), but do not require low latency. Data services usually don't require either low latency or low jitter.

No one protocol is optimum for all types of data, but it is possible to combine elements of more than one protocol to produce a communications system that is reasonably good for all needs. In the downstream direction, data originates (or at least is codified) at the headend. Two methods of transmitting the data are frequency- and time-division multiplexing. Frequency division multiplexing (FDM) is used in NTSC transmission, in which every signal is modulated onto its own carrier at a different frequency. In time division multiplexing (TDM), different signals (or data streams) are combined on one carrier by transmitting one for a time, then transmitting another. This latter method can be more efficient in terms of the total bandwidth required. If more than one service is required, such as suggested in the NIM (figure 1), then TDM allows only one receiver to be used for multiple services, reducing cost.

In the upstream direction, FDM can be used, but is usually not chosen due to the relatively inefficient use of the upstream bandwidth. TDM is not used, as there is no point at which all upstream traffic is brought together to be formatted into a single data stream. Most of the protocols we have seen recently have involved in some way, reserved time division multiple access (R-TDMA. Each upstream talker is assigned a time slot during which he can transmit upstream. The time slots are assigned from the headend. In some cases, the time slots are fixed, with one talker being assigned exclusive use of a certain time slot. In other cases the time slot is assigned to those talkers who have indicated that they have something to say.

Other methods have been proposed to return signals upstream from multiple talkers. The most talked-about is probably code division multiple frequency modulation, a form of spread spectrum communications. Each talker is assigned the same wide frequency band and each talks at the same time. The difference is that different talkers use different spectrum spreading sequences. Knowing each talker's spreading sequence allows one to "take apart" the individual signals at the headend.

Fixed frequencies are usually not assigned for all time in the return band. At least two reasons exist for not assigning fixed frequencies. The nature of interference in the return spectrum is such that a frequency may be good at some times and not at others. Also, in order to efficiently use the return spectrum, it is necessary to allocate bandwidth only when a particular service needs it. For example, one would not like to dedicate bandwidth to a remote meter reading service that needed a few milliseconds each month to read meters in each home. Rather, one would like to use that bandwidth for another service when it is not needed for meter reading.

## THE SPECTRUM MANAGER

These concerns may be addressed by employing the concept of dynamic spectrum management. This means that there exists some mechanism that knows what use is being made of the return spectrum (and downstream spectrum, too), and is able to assign different services to parts of the spectrum appropriate for their immediate needs and the condition of the return spectrum. This mechanism has been called a "spectrum manager," or a "spectrum management agent." Generally, it includes some sort of ability to assess the quality of the return spectrum, knowledge of the demands being placed on the spectrum, and the ability of different services to control their functionality. The spectrum manager is able to command different services to operate together to maximize efficiency in the return band.

The spectrum manager typically includes some sort of detector to assess the return spectrum quality by frequency, and is able to tell various services what frequency and power levels to use, and may be able to change modulation methods. A computer is assigned the task of the spectrum manager. It will allow parameter monitoring and automatic or manual adjustment. In more advanced systems, it will interface with other levels of a hierarchical system that monitors all aspects of a cable system's performance, and allows rapid pin pointing of faults. Such higher level systems are known as operational support systems (OSS).

## LOGICAL ACCESS TO FREQUENCY MANAGEMENT

The following discussion describes how one logically controls the system using a frequency manager. This does not represent the physical data flow. The terminology is borrowed from the telephone industry, which has been using systems such as this for a long time.

Figure 3 illustrates the logical method by which monitoring and control of the system is achieved. A cable television system is subdivided into a number of **logical nodes.** In the limiting case the logical nodes are the same as the fiber nodes. In the interest of economy, it is likely that most systems will want to combine several nodes at the headend, until adequate revenue streams exist to support the additional equipment needed to serve each fiber node individually. Thus, for the present purposes we define a logical node that is equal to the fiber nodes that are supplied by one

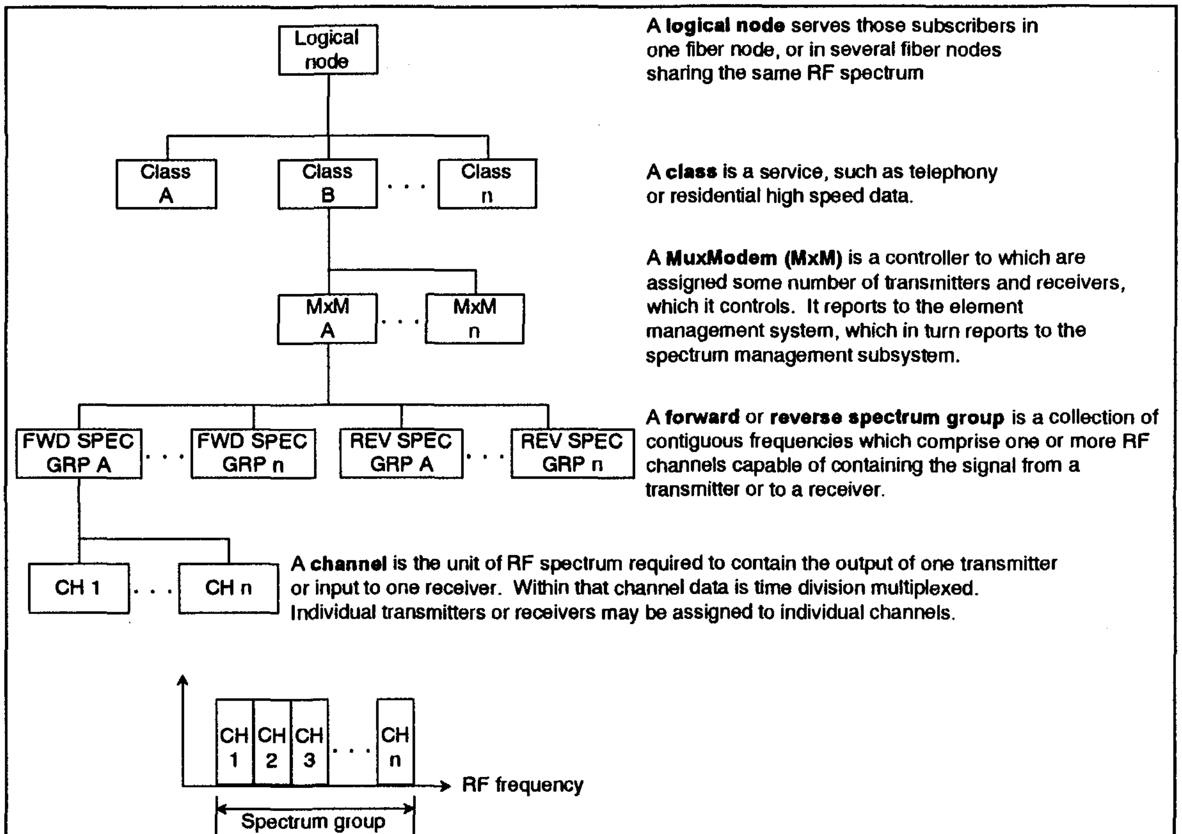


**Figure 3. Logical Interfaces**

fiber transmitter and which use one fiber receiver for the return path. This grouping is characterized by the need to coordinate all frequencies within the node, but no need to coordinate with other nodes.

**Classes** refer to services that need, in some way, differing characteristics in the data communication system. At the moment the following classes are defined.

1. Telephony.
2. Residential high speed data.
3. Switched digital services.
4. Utility communications services (including security monitoring).
5. Converter return data.
6. Integrated multimedia services.
7. Transponders (conventional cable status monitoring of distribution plant).
8. Invalid (a method to mark a certain RF bandwidth which has been identified as being unusable, for example due to ingress. The identification may be either automatic or manual.)
9. Unavailable (RF bandwidth reserved for future use, or occupied by services not covered by the system).

The **MuxModem** is the controller for all modems serving a common logical node. More than one MuxModem may serve the same logical node if adequate frequency spectrum is available. The MuxModem microscopically controls the assignment of modems to various spectrum groups and channels, within the limitations assigned to it by the element manager. In addition, it handles other aspects of modem operation. It communicates down to the individual transmitters and receivers, and up to the element manager.

The **forward** and **reverse spectrum groups** are a set of contiguous RF frequency bands which contain one or more RF channels. Each channel is capable of containing one RF path to or from a logical node. If a particular class is eligible to communicate on non contiguous bands, those bands must be set up as separate spectrum groups.

Each spectrum group is made up of one or more **RF channels**. One RF channel is wide enough to contain one transmission from the headend or from the home. An RF channel may not be subdivided within a class, as it is the minimum entity that can carry data. In different classes, RF channels are of different width (the width is expressed in KHz or MHz of RF spectrum). Thus, in one class, a spectrum group that is eight MHz wide may contain 4 RF channels, each two MHz wide. In another class, the same spectrum group may contain 8 RF channels, each one MHz wide. Within a spectrum group, all RF channels are of the same width.

Within each channel, a number of subscriber services are time division multiplexed, to enable service to many subscribers at once within that RF channel. However, since the spectrum cannot be broken down below the RF channel level, the spectrum manager doesn't care about how the channel is used.[iv] The only metric of use to the spectrum manager is the percent utilization of the RF channel. If all RF channels available to a particular MuxModem are approaching their capacity, then the spectrum manager may have to find additional RF channels in spectrum groups not presently available to the MuxModem.

## UPSTREAM SIGNALS NEEDING TO BE CONTROLLED BY THE SPECTRUM MANAGER

The table on the following pages lists the services needing to be controlled by a spectrum control system, and the parameters that would need to be controlled. This is an early attempt to catalog the return services we expect to see in the next few years, but is not by any means a comprehensive list. It should serve to illustrate the subject, however. In addition to the management shown, we expect that downstream equipment will also be controlled, but in the interest of needed brevity, we omit it here. The table is discussed at its end.

**Components needing control by Spectrum manager**

| Component | Description | Characteristics | Tuning Res. | Monitor | Control |
|---|---|---|---|---|---|
| 1. Upstream data. (Includes transmitter at headend and receiver in field) | Existing upstream data for pay per view | FSK or BPSK, variable bit rates. Usually a combination of polling and ALOHA protocols | Varies by manufacturer. | Manual, cannot monitor | Must mask out manually. |
| 2. Point to point modem, upstream receiver (headend) | Nailed up connections, continuous communications. | 1.544-6.176 Mb/s above data link layer. QPSK, < 1 MHz to < 4 MHz occupied. Symmetrical data rate. Low jitter. | 0.25 MHz to mesh with STD, HRC, IRC assignments. | -Receive frequency.<br>-RSSI.<br>-Eye closure. | -Receive frequency.<br>-Expected signal level. |
| 2a. Point to point modem, upstream transmitter (in field) | see 2 | see 2 | see 2 | -Receive frequency.<br>-Transmit frequency.<br>-Output attenuator. | -Receive frequency.<br>-Transmit frequency.<br>-Output attenuator. |
| 3. Point to multipoint upstream receiver for constant rate services (headend) | Switched connections, telephony and related services | 1.544 Mb/s above data link layer. QPSK, ~1 MHz occupied. Symmetrical data rate. Low jitter. | 0.25 MHz to mesh with STD, HRC, IRC assignments. | -Receive frequency.<br>-Expected signal level.<br>-RSSI.<br>-Eye closure.<br>-Utilization. | -Receive frequency.<br>-Expected signal level. |
| 3a. Point to multipoint modems for constant rate services, field unit | see 3 | see 3 | see 3 | -Transmit frequency.<br>-Receive frequency.<br>-Output attenuator.<br>-RSSI.<br>-Eye closure.<br>-Unsuccessful attempts. | -Transmit frequency.<br>-Receive frequency.<br>-Output attenuator. |

| Component | Description | Characteristics | Tuning Res. | Monitor | Control |
|---|---|---|---|---|---|
| 4. Point to multipoint upstream receiver for variable rate services (headend). | Most data services (computer communications). Data delivery may include considerable jitter. | ~1.544 Mb/s above data link layer (may want to reduce), QPSK, ~2 MHz occupied. | 0.25 MHz to mesh with STD, HRC, IRC assignments. | -Receive frequency.<br>-Expected signal level.<br>-RSSI.<br>-Eye closure.<br>-Utilization | -Receive frequency.<br>-Expected signal level. |
| 4a. Point to multipoint modem for variable rate services (field unit). | see 4 | see 4 | see 4 | -Transmit frequency.<br>-Receive frequency.<br>-Output attenuator.<br>-RSSI.<br>-Eye closure.<br>-Unsuccessful attempts. | -Transmit frequency.<br>-Receive frequency.<br>-Output attenuator. |
| 5. Telemetry receiver (headend) | Low bit rate data. For remote meter reading (poling), security (slotted ALOHA?), etc. | ~2400 b/s FSK (may be faster but stay within channel) | ~50 KHz occupied bandwidth. | -Receive frequency.<br>-Expected signal level.<br>-RSSI.<br>-Eye closure.<br>-Utilization | -Receive frequency.<br>-Expected signal level. |
| 5a. Telemetry modem (field unit) | see 5 | see 5 | see 5 | -Transmit frequency.<br>-Receive frequency.<br>-Output attenuator.<br>-RSSI.<br>-Eye closure.<br>-Unsuccessful attempts. | -Transmit frequency.<br>-Receive frequency.<br>-Output attenuator. |

General notes to the table:

1. RSSI = received signal strength indicator. This is a voltage (or word) proportional to received signal level.
2. Output attenuator on a transmitter is an indirect indication of output level. We assume that in the majority of cases transmitter manufacturers will supply this information as an output level indicator, but we must have more information to derive output power. Since output power is important, we assume that manufacturers will supply a tabulation for each piece of equipment, relating power level to attenuator setting. The accuracy of the table will be only moderate, however, and a deviation of several decibels can be expected. This is important primarily in the interpretation of data.
3. "Unsuccessful attempts" is a measure of the number of times an attempt was made to access the system without success. The criteria and reporting format are not determined yet.
4. Eye closure is taken as a general measure of received signal quality. Other metrics could be used, but this seems to be a good choice.
5. "Expected signal level" in headend receivers refers to the need in such receivers to tell them what the proper signal level to which they should control the upstream transmitters. This will vary with headend configuration. A manual, or better, automatic method of setting this must be developed.
6. Other monitoring and control functions may apply to some pieces of equipment. Only those which seem related to spectrum management are shown here.

Discussion of elements of the table

1. The first entry is represented by RF IPPV systems in use today. These systems are not capable of being controlled by a higher entity, and therefore would be omitted from the spectrum manager (class 9 above).

2. Point to point modems are those intended for use between two points. For example, we are seeing new interest in T1 modems for use between business locations. These modems operate at 1.544 Mbps and can carry 24 voice channels or other combinations of voice and data in 64 Kb increments. For example, a medium size business may lease a T1 line to get all of its telephone traffic to a telephone central office or to the point of presence of a long distance carrier. This traffic is carried by the telephone company today, but is capable of being carried by cable, possibly at very competitive rates.

The service is characterized by continuous communications between two points. Communications that is always carried out between the same points is referred to as a "nailed up circuit," as opposed to a switched circuit. We refer here to a field modem and a headend modem, which are identical except for complementary frequency plans. The headend receiver is expected to report to the spectrum manager the following items.

a. Receive frequency - confirmation of the frequency to which it is tuned.
b. RSSI - received signal strength indication is a measure of the received signal strength. This information is used by the spectrum manager to control the field modem transmitter. Because the system gain from the field location to the headend is not precisely known, it is necessary to measure the signal level received at the headend, and to remotely control the filed transmit power to make the received signal level correct.
c. Eye closure - a measure of the quality of the received signal. This data tells the spectrum manager whether anything is interfering with the received signal, when the interference is low enough so as not to corrupt the data. The spectrum manager may then make a decision to raise signal levels above the nominal target, or to move upstream transmission to a new frequency.

In turn, the spectrum manager controls the received frequency and the expected signal level. The received frequency must be changed if the manager determines that the interference on the present frequency is too high for reliable communications at the highest signal level permitted, so that a frequency change is indicated. The headend receiver must be informed of the expected signal level because the loss in the headend is not generally constant. System design will dictate the target signal level at the headend from the field transmitter, based on reverse amplifier and transmitter signal levels. In general, from the input to the headend to the headend receiver, some gain or loss is expected, based on individual headend design. During system set-up, the expected signal level is computed, and the receiver so informed.

2a. The point to point upstream modem communicates with the receiver in 2 above. It reports its receive and transmit frequencies (for confirmation) and its output attenuator setting. The output attenuator setting is a measure of transmitter power output, which we choose not to call "output power" for technical reasons. It is a measure of how much power the transmitter is putting out, a metric useful for diagnosing upstream transmission problems. The spectrum manager controls the transmit and received frequencies (so it can change them based on current conditions) and the transmit attenuator (power).

3. Point to multipoint communications for constant rate services are particularly suited for telephony and related services requiring constant symmetrical data rates. The monitoring and control strategy is much the same as shown above, with possible the addition of a report on utilization and unsuccessful attempts. These additional items are particularly germane if contention is a part of the system. That is, if the number of possible users exceeds the number capable of simultaneous service. The utilization tells the spectrum manager how much excess capacity exists on the frequency, in case more subscribers need to be placed there. The number of unsuccessful attempts ("busy signals") tells the manager when more capacity is needed.

4. Point to multipoint communications for variable data rate services represent most computer communications which might be carried on cable. For example, cable is in a good position to offer communications for the commercial on-line services such as Prodigy, America On Line and Compuserve, and for Internet access. These services are characterized by very bursty traffic, with long pauses in between. Furthermore, the services are highly asymmetrical, with more information being sent downstream than upstream. For illustration, we assume an upstream data rate of 1.544 Mbps. The table from which this was extracted showed a downstream rate of 10 Mbps. The monitoring and control issues are the same as for the constant rate services shown above.

5. Telemetry services include energy management and home control. They have many of the same characteristics as point to multipoint communications, with the exception of lower bit rate and possibly slightly more symmetrical data requirements. In fact, these services could be combined with the point to multipoint, except that less efficient but lower cost modulation formats may be used.

## CONCLUSION

The above examples show some of the monitoring and control tasks that are likely to be accomplished by a spectrum manager, related to services involving upstream communications. We have concentrated on upstream communications because this is likely to be the portion of the spectrum in which we find the most congestion in the next few years, so it is the place to put most of the effort to improve the utilization. One way to improve utilization and service availability is to provide a spectrum management system, which automatically monitors usage of the spectrum and which can move services around as needed to provide the most efficient utilization and the maximum availability.

## ACKNOWLEDGMENTS

The author is grateful for the education provided by Jack Terry of Northern Telecom, Tom Engdahl of Digital Video (division of Antec), Michael Pritz of Antec, and Jim Stratigos of Wireless Communications Consultants. Shellie Rosser

of Antec surveyed operators concerning their expectations of revenue from various services. Valuable information and insight were provided by Tom Williams of CableLabs.

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[i] Real diplex filters are, of course, much more complex than the simple representations shown here.
[ii] Terry, J. B., "Challenges and Solutions in the Introduction of Digital Services in Home Coax Wiring," *NCTA Technical Papers*, 1995.
[iii] Available from the Electronic Industries Association Consumer Electronics Group, 2500 Wilson Blvd, Arlington, VA., (703) 907-9600.
[iv] A system employing time division multiplexing or some other sharing arrangement within a channel would arbitrate within itself, the multiplexing within that system. The spectrum management would only come into play when the system was running out of capacity and needed more spectrum.

# MINIMAL BANDWIDTH UPGRADES

**Ted E. Hartson**
**Post Newsweek Cable**

**Abstract**

*The operational bandwidth of a cable television system is normally limited to something at or near it's designed bandwidth. The designed bandwidth of a system frequently is more perceived than actual. The ability to accurately analyze the factors at play in an existing system may provide the understanding necessary to embark on a very cost effective bandwidth upgrade.*

*A clear understanding of how these factors interrelate and their economic burden can frequently result in the planning and execution of a project that provides bandwidth enhancement and additional revenue.*

## HURDLES TO HAPPINESS

Minimal expansion of bandwidth in a system depends on the condition of several elements in the plant. These elements may be thought of a hurdles. Each hurdle must be cleared or the project can not be completed. The cost in work hours and dollars associated with each hurdle must also be clearly understood. If every hurdle is high and it's associated cost large, then the system may not be a candidate for "MBUing". Accurate investigation, costing, and planning is essential or you may find yourself trapped. The major subsections of a system are: 1) Trunk Amplification, 2) Bridgers and Line Extenders, 3) Taps and Passives, 4) Cable.

The amount of effort which must be expended in each of these areas is a direct result of the overall bandwidth expansion desired. Beyond a certain expansion bandwidth, which is different in each system, the costs and effort will soar.

A way of maintaining perspective in exploring MBU options is to equate the impact of the project as cost per channel per subscriber. In this way, pushing beyond the economic knee will be reflected in raising costs per channel per subscriber . A separate important analysis not provided here is the revenvue production capability of newly created channels.

## WHAT TO EXPECT

At the end of this paper we will look at some actual MBU information on real systems. MBUs adding 6 to 10 channels in systems with 300 to 450 MHz as the current upper band limits are reasonable expectations.

## HURDLE ONE

The bandwidth and number of trunk amplifiers in cascade is the first concern. Many systems have trunk modules that are capable of full performance above their specified upper frequency limit. The principal band limiting element in a trunk station is the equalizer. Selection of an equalizer with a higher top frequency will solve this problem. Before you rush out for a bag of new equalizers, let's look at the other problems. Most trunk/bridger stations pass AC power. This is accomplished by a series of several power passing chokes and bypass capacitors that provide a directable power path through the amplifier which is independent of the RF function(s). Power passing chokes are selected by the designer to be highly reactive within the design frequency range. When properly sized they are essentially invisible to the RF path. Once you explore the band beyond the upper design limit you run the risk of entering a region where the power passing chokes absorb RF energy and hence influence Gain/Frequency response. This "choke notch" when encountered will be pronounced and set a practical absolute limit to the bandwidth that can pass through a station. Some hard core RF Engineers may talk about moving this notch by bending or "knifing" coils. It can be done, but, it's well beyond the scope of this paper. A couple of other factors also get in the way: As you go up in frequency the attenuation of coaxial cable increases. The expected attenuation between stations must be accounted for. It may range from an inconsequential amount to several dB. Maintaining operational levels at higher frequencies also becomes a problem when the gain of the station drops off a few tenths of a dB and internal passives manifest a little extraordinary loss. These factors together eat into residual operating margins. Careful bench testing of a sample of stations at the newly proposed upper frequency and actual inventory of the trunk stations in the field to ascertain pad values, (that's gain you can use), will address these concerns.

Don't plan on faking any of this. Careful deliberate steps are the difference between success and failure in MBU's.

Before we leave trunk amplifiers, a couple of operational comments. Most systems are noise based, this is, they are positioned within their operational level window toward the noise side. This is because until recently we have lacked high quality low cost field equipment for making cross modulation and composite triple beat measurements. (*such as the Hewlett Packard 8591C*). This omission resulted in "staying as far away as possible" from coherent distortions. The reason is simple! The onset of coherent distortion in subjective analysis is abrupt and most observers will see it within a couple of dBs of one another. On the other hand, noise is more gradual and highly subjective. It was easier to sell noise than beats.

Today, their is no excuse for not being well positioned within the operating range; this generally happens through raising the levels. It may be counter intuitive to think about adding channels and raising levels, but it may turn out that way!

Finally, if you have applied any glass to your system to reduce cascades and enhance reliability the reduction of cascades not only relaxes noise/distraction objectives, but also relaxes gain/frequency considerations in that the cumulative impact is lessened.

## HURDLE TWO

The performance of the bridger and the line extenders in a distribution leg set not only the performance but the tap levels, as well, hence drop levels for the individual subscribers. The impact of changing levels must be thought through as to how it relates to drop levels and the Commissions (FCC 76.605) requirements. Changing pads and equalizers to reach a higher bandwidth is a good investment, dropping in higher gain modules may be a good investment. Changing out line extender stations may not make economic sense for an MBU.

When the design bandwidth of a bridger or a line extender is slightly exceeded, the signals will start to fall off on the higher channels, that right where you need it to overcome tap losses that are also increasing as they approach their performance limits. A practical way of quantifying the impact of many of these factors is to test a selected sampled that picks up a few representative distribution legs.

Detailed testing including the net impact on distribution end performance will answer these questions.

## HURDLE THREE

Taps and passives while they look and work about the same, their impact on the cost of an MBU can be significant. If passives restrict bandwidth because of extraordinary loss or more likely power passing choke notches, they can simply be replaced. Passives are cheap and about a dozen or less per plant mile. Taps on the other hand are everywhere; underground, down everybody's street and up your alley. Taps generally degrade gracefully and pushing bandwidth by perhaps 10% (330 from 300 or 440 from 400) should not present a problem. Beware, however, the dreaded power passing chokes. Some taps are into the power passing choke notch at even 10% above the upper frequency limit. this is an area in which good attention to detail is necessary.

Different taps within from the same manufacturer and same series may behave quite differently. If a few values are found to be a problem, they can probably be changed out economically, however reaching every tap in a system to achieve an MBU makes no sense. An inventory of the type and value of taps used within the system and their performance at the MBU frequencies will provide the needed information. All taps have power choke notches, passive sweep testing of a tap cascade in the field is a good way to accelerate this understanding. Testing of a tap or two on the bench may provide misleading data. Generally, all devices of the same manufacturer and tap values will behave the same. As mentioned earlier, a few problematical taps can be exchanged in the pursuit of an MBU.

## HURDLE FOUR

The coaxial cable used in a system frequently is the highest hurdle. Cable can be a disappointment in two ways: 1) It's original design performance fall short of a new expectation, or, 2) It is defective and fails to meet it's original performance specification.

If you think taps are everywhere how about cable: "Gee, that's why they call it Cable TV!" Occasionally one may encounter a style cable that will express some unusual behavior, in a frequency range. This is generally due to mechanical properties of the cable, repetitive structures or damage in a cable can set up a circumstance where a notch or "suck out" can occur. These properties can be tested for by sweeping. The writer has had some experience with a cable no longer in manufacture called Seal-a-Matic. This cable had an outer conductor made of aluminum foil and over folded against itself logitiutionally. While this cable was blamed for many sins, including signal leakage, the reality was that with good connectors is was and is capable of operation at 450 MHz.

The message here is that older cable should not be rejected out of hand for expanded bandwidth projects. There is no substitute for sample testing. A more frequent problem is one of cable that is bad for a variety of reasons and all the margin is used up; *even at the old bandwidth*. Sample testing and extrapolating results into the overall project will generally prove that only some fraction of the total miles of a plant suffers from bad cable and at some price it can found and fixed.

## MBU POLITICS

Bringing together a successful MBU or bigger still a series of them can result in adding channels to existing systems. These channels can generate new revenue sources, as well as taking permissive rate increases through channel expansion. The prospects of adding channels through MBU can easily be misunderstood: *MBU is not an alternative to system rebuild, it is an interim step that provides more channels sooner. MBU is not risking arrest. It is not speeding to do 330 MHz in a 300 MHz zone. MBU is not bad engineering, in fact it is just the opposite, sound proven engineering is what makes it work.*

The close interchange of ideas, and expectations between management and engineering is absolutely necessary to make MBU successful. The overall number of channels than can be achieved is dependant on a number of variables. While some systems may be similar in their MBU needs, no two will be exactly alike. Trying to force too aggressive an MBU into a system will result in high costs per channel. Alternatively, adding channels with only minimum preparation runs the risk of poor performance, problems with subscribers and /or the Commission at proof time.

Here is some actual MBU information and a broad estimate of the cost of implementation in various type systems. You may find this information helpful in ranking a prospective MBU project.

TYPICAL MBU PROJECTS

| System | System Current Upper Limit | MBU Upper Band | Cost/ Channel Sub | Notes |
|---|---|---|---|---|
| A | 300 | 400 | $2.83 | Small, |
| B | 300 | 400 | $3.03 | similar |
| C | 300 | 400 | $2.50 | in size |
| D | 300 | 400 | $2.56 | and age. |
| E | 400 | 460 | $9.00 | |
| F | 400 | 450 | $3.40 | |
| G | 340 | 380 | $7.67 | Previously expanded from 300 340 MHz. |
| H | 270 | 330 | $3.40 | |
| I | 300 | 330 | $5.24 | |
| J | 300 | 330 | $6.24 | |
| K | 300 | 330 | $5.21 | |
| L | 400 | 450 | $2.77 | |
| M | 300 | 366 | $23.62* | Extensive .412 Dist. |
| N | 330 | 330 | $15.28* | .412 Dist. Band limited taps. |
| O | 270 | 330 | $14.95* | Needs taps & cable. |
| P | 300 | 330 | $16.22* | Needs taps & cable. |
| Q | 330 | 450 | $6.25 | 450 L/E & taps in place. |

Average unweighed cost per channel per subscriber, is $4.62 excluding projects not undertaken shown by * above.

## CONCLUSION

Points to ponder:

- o Every system is different, the price of an MBU varies.

- o The number of channels that can be economically obtained varies.

- o Not every system is an MBU candidate.

- o Some aspects of the capital and labor expenditures of an MBU may be useful in a subsequent rebuild.

- o MBU projects which can be finished quickly provide revenue that is otherwise lost or delayed until a rebuild.

Additional channel capacity may be falow in your system, why not put it to work soon?

# Noise Considerations in Coaxial Cable Systems

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## ABSTRACT

*Coaxial cables will be used to transport analog and digital signals to and from the home. The full spectrum of the coaxial system, from 5 MHz to 1 GHz will be utilized for the delivery of video, voice and data. The natural shielding characteristics of coaxial cable will play a primary role in helping to maintain signal integrity and ensure error free transmission. This paper addresses the shielding performance of coaxial cable and its related components. This is accomplished through a discussion of leakage mechanisms and laboratory shielding tests.*

## INTRODUCTION

Signal distortion due to noise ingress in a coaxial cable system has generated considerable concern and discussion. In the past, shielding was primarily an egress (signal leakage) issue. This was driven by Part 76 of the Federal Communications Commissions (FCC) Rules and Regulations. Radiated emissions are required to be measured and must meet the following limits as shown in Table 1.

| Freq., MHz | Signal Strength Limit, µV/m | Distance, ft. |
|---|---|---|
| $f \leq 54$ | 15 | 100 |
| $54 < f \leq 216$ | 20 | 10 |
| $216 < f$ | 15 | 100 |

**TABLE 1.** - FCC Limits

In most cases, if the conditions were satisfied, then the shielding integrity was sufficient enough to alleviate any noise ingress issues. The appropriate question becomes, whether or not meeting the FCC leakage requirements also satisfies the needs of a digitally based two-way interactive communications systems. To try and answer this question, we need to discuss what the communications system will look like.

It appears that the network will end up being a fiber-star coax-bus extending from a fiber ring or virtual ring architecture. The coax plant will then consist of coax feeders and distribution cable extending from a node to the tap at the curb with amplifiers in-between. From the tap, drop cable, connectors and splitters will be used to connect the customer to the coax-bus. It is obvious that the coaxial cable and its related components will be in contact with numerous noise sources. Sources of noise could be anything from indoor microwave ovens, computers and appliances to outdoor ignition switches, power devices and terrestrial broadcast antennas. The coaxial cable system, therefore, has the unfortunate potential of behaving as a distributed antenna, picking up unwanted signals from the home and throughout the neighborhood.

It then becomes obvious that an understanding of cable shielding would be beneficial, when designing these networks, so as to help ensure acceptable performance. The shielding performance of an outer conductor, whether it is for a cable, connector, tap or splitter - is a function of material, design/construction, installation technique and field conditions as well as frequency of operation.

The three mechanisms which can contribute to the overall shielding performance are absorption (diffusion), aperture leakage (coupling) and wave reflection (mismatch). Each of these mechanisms are a function of frequency and are described in greater detail in the next section.

It appears that the frequency allocations will be re-formatted, in order to meet the network needs. Table 2 gives a representation of what a typical frequency allocation might look like.

| Freq., MHz | Direction | Format |
|---|---|---|
| 5-40 | Upstream | Digital |
| 54-550 | Downstream | Analog |
| 550-750 | Downstream | Digital |
| 750-1000 | Two-Way | Digital |

**TABLE 2.** - Typical Frequency Allocation

The other obvious consideration is the carrier-to-noise requirements needed for each particular service. Digital telecommunications systems operate better than $1 \times 10^{-10}$ Bit Error Rate (BER), where $1 \times 10^{-3}$ BER is rendered unusable.[1] For the upstream communications, a carrier-to-noise ratio of 15.8 dB for QPSK and 23.3 dB for 16 Quadrature Amplitude Modulation (QAM) is required. For downstream QAM modulation, a carrier-to-noise ratio of 25 dB is required for 64 QAM and 16 dB for 16 QAM.[2]

## SHIELDING MECHANISMS

The metallic outer conductor of a coaxial device creates a natural isolation between the internal communication signal and the external environment. The internal electrical and magnetic fields are contained within the first few thousands of an inch. The fields and corresponding currents are attenuated by the factor $e^{-\alpha z}$. The skin depth is defined as the distance the energy must travel in order for its amplitude to decay exponentially, $e^{-1}$, or 37 percent of its initial value.[3] (See Figure 1) For aluminum, one skin depth is 0.001 inches at 10 MHz and 0.0001 inches at 1 GHz. The skin depth can then be used to calculate the absorption loss through a conductive shield (Fig. 1 and 2).[3] This equates to about 9 dB of shielding for one skin depth, therefore, a 0.010 inch shield would have approximately 90 dB of shielding at 10 MHz.

The shield absorption term does not account for any signal leakage that occurs through holes or apertures in the outer conductor. Aperture leakage is usually the dominant factor in cable shielding and is a complicated phenomenon. Figure 3a illustrates the effect that aperture orientation has on shielding effectiveness. Current flow in a coaxial shield is intended to be longitudinal along the length of the outer conductor. An opening in the shield causes a disruption in the flow of electrons as they have to travel around it. It is this disruption of current flow that causes signal leakage or coupling through the outer conductor. Figure 3b illustrates the approximated relationship among width of opening, frequency of operation and shielding effectiveness. The equation illustrates that the shielding effectiveness is inversely related to the signal frequency and circumferential slot width. For example, a 0.010 in. slot placed in the outer conductor of a trunk cable will have a shielding effectiveness of 94 dB at 10 MHz and will degrade to 54 dB at 1 GHz.

Reflective losses will also occur at the outer conductor interface. This is typically a noise ingress issue. The electromagnetic wave propagation of the signal inside a coaxial device is along the length of the cable. Since the angle of incidence of the wave propagation to the outer conductor is zero degrees, there is essentially no reflective component. This is not the case for external fields which can ingress into the coax. External fields can impinge on the outer conductor at an unlimited number of angles. Reflections occur as the incident wave which is traveling through a high impedance medium (free space $Z_{free}$ = 377 Ohms) meets a low impedance medium ($Z_{aluminum} = 4.67 \times 10^{-7} \times f^{1/2}$). This generates a reflected signal who's energy is proportional to the ratio of free space impedance to the shield impedance. This is shown in the equation below and a plot can be found in Figure 2

$$\text{S.E. Reflection} = 20 \cdot \log \frac{377}{4 \cdot (4.67 \cdot 10^{-7} \cdot f^{1/2})} \text{ dB}$$

## S.E. MEASUREMENT TECHNIQUE

Due to the complexity of the leakage component found within the coaxial cable network, computation is not possible, therefore, shielding effectiveness must be determined by measurement.

The most obvious way to do this and the method which most closely simulates actual performance of the leakage component is to measure the ingress energy within the component resulting from a known external electromagnetic field, or conversely to measure the egress (energy emitted) resulting from a known energy level within the component under test. The systems engineer would then be able to use this information to determine shielding effectiveness specifications for the individual components as well as predict the shielding effectiveness of the entire system with high reliability.

Traditionally, within the CATV industry, the parameter used to measure the penetration of electromagnetic energy through a cable's shield is the transfer impedance (Zt) and is measured using the triaxial chamber test setup. The transfer impedance has been used to give a comparison between cable shields and is quite useful for this purpose. However, its ability to determine the cable's shielding effectiveness (i.e., radiated field strengths) is questionable. Other limitations of the triaxial chamber setup include only being able to measure cables and not connectors, splitters and other components. The industry needs:

- An industry test standard where acceptance standards can be applied globally, measured uniformly and compared directly.
- Meaningful measurements that correlate to Open Area Test Site (OATS) measurements.
- Capabilities which include repeatability and accuracy. Ability to measure shielding effectiveness of cables, connectors, taps and other components (not limited by geometry).

### GTEM Cell

A new test cell which offers the above mentioned industry needs where either emissions or immunity testing can be performed is the Gigahertz Transverse Electromagnetic Cell, the GTEM cell and is illustrated in Figure 4. The GTEM represents a significant advancement in state-of-the-art EMC testing. As part of a measurement system, it allows more types of testing to be done in less time while improving accuracy and repeatability of results.

The GTEM cell is an offshoot of the TEM cell, as developed at the National Institute of Science and Technology (NIST). It is a section of 50 ohm transmission line with a unique geometry. At the input a normal 50 ohm coaxial transmission line is transformed into a rectangular cross section. The cell is flared along the longitudinal axis to increase the cross sectional dimensions of the transmission line. The septum, or center conductor, is transformed from a round cross section to a flat wide conductor, located well above the center of the cell. This maintains the 50 ohm characteristic impedance, while increasing the volume of the cell under the septum, allowing a larger test volume. The GTEM can perform measurements from DC to 1 GHz and above and emissions testing has been found to correlate with open area test site measurements. Connectors and other components can also be tested for emissions or immunity, allowing their shielding performance to be measured.

The GTEM has been used to measure the shielding effectiveness of various leakage components and the data is presented in this paper. Refer to Figure 4 for the GTEM Test Setup. The shielding effectiveness parameter is defined as the level difference, in dB, between the signal level inside the Device Under Test (DUT) to the signal level immediately outside the DUT, measured with a 0 dB gain 1/2 wave dipole tuned antenna.

## S.E. OF COAXIAL CABLE COMPONENTS

This section details the results of the S.E. measurements for the various coaxial cable components. Measurements were performed using the GTEM cell, as described in the previous section (Refer to Figure 4 for measurement setup). Table 3 gives a summary of the components and their related plots.

**S.E. Plots**

| Description of Component | Figure No. |
|---|---|
| Longitudinal Slot | 5 |
| Circumferential Slot | 6 |
| Expansion Loop Failure | 7 |
| Connectors (T&D) | 8 |
| Splitters (Drop Cable) | 9 |
| Taps (4-port) | 10 |
| Connectors (Drop Cable) | 11 |
| Drop Cable (Unflexed) | 12 |
| Drop Cable (Flexed) | 13 |

**TABLE 3.** - S.E. Plots

### Slots

It can be seen from the S.E. plots in Figures 5 and 6, that the circumferential slot has much greater RF leakage than the longitudinal slot. This is because a slot placed circumferentially, normal to the direction of current flow, abrupts a greater portion of the current densities, thus causing greater leakage.

Figure 7 shows the S.E. of an expansion loop failure for a solid outer conductor cable design, i.e, T&D coaxial cable. Failure at the expansion loop is a result of work hardening of the coaxial cable's outer conductor, due to the continuous flexing that takes place. Failure occurs in the center of the expansion loop, where a circumferential break in the outer conductor occurs. Because the expansion loop failure represents a circumferetial slot, the circumferential slot and expansion loop S.E. plots are very similar (Figures 6 and 7, respectively).

### Connectors (T&D)

The majority of signal leakage sources within a coaxial cable system are caused by the connector/cable termination. Leakage is typically due to the loosening of the connector/cable outer fitting, effectively leaving a gap in the outer conductor of the coaxial cable. It is through this circumferential slot that electromagnetic energy reaches the exterior region.

From looking at Figure 8, it can be seen that T&D connectors, when installed properly, have very good S.E. performance, approximately 125 dB average and 115 dB minimum shielding effectiveness. However, with the slightest loosening of the connector, the S.E. is seen to decrease considerable, from 115 dB to 53 dB minimum S.E.

Taps

Figure 10 shows the S.E. of a 14 dB, four port tap, with and without its ports terminated. The tap is seen to have acceptable S.E. performance, as long as its unused ports are terminated properly. The unused ports, when left unterminated, are seen to decrease the average S.E. from 135 dB to 110 dB.

Splitters

Various constructions of drop cable splitters can be found within the industry which have different levels of S.E. performance. Figure 9, for example, shows the S.E. performance of a splitter having a soldered back vs a splitter having a glued back. The two splitter designs are seen to have very different S.E. performance, 106 dB vs 43 dB respectively.

Connectors (Drop Cable)

Figure 11 shows the S.E. measurements for various types of drop cable connectors. There was found to be little difference among the crimp, compression and push-on connectors.

Drop Cable (Unflexed and Flexed)

Figure 12 shows a S.E. comparison among standard, tri and quad-shield constructions as well as non-bonded standard shield. Results were as expected, showing that cables with more shielding have greater S.E. performance.

Flexure is used to simulate aging which includes cyclic stress from temperature changes, wind and ice loading. Flexure has greatest effect on the tape and does not significantly degrade the braid's performance.[3]

Figure 13 demonstrates the S.E. of the drop cable constructions after flexure, all samples being equally conditioned.

Conclusion

From looking at the S.E. performance of the various coaxial cable components, it becomes evident that the systems engineer should specify the S.E. requirements of the individual components which are to be used within the coaxial cable system. This will then help to ensure that the system, after being built, will have adequate S.E. performance, initially and also after being aged, in order to reduce maintenance.

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## ACKNOWLEDGMENTS

The authors would like to thank Mr. M. Pennell and Ms. T. Benson for their support.

## Shielding Effectiveness Through Absorption Loss (Skin Effect)

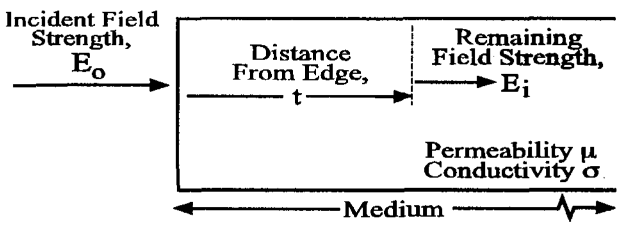
**Shielding Effectiveness:**

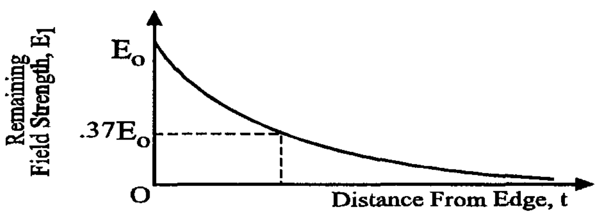
$$S.E. = 20 \log\left(\frac{E_o}{E_i}\right) = 20 \log\left(\frac{H_o}{H_i}\right) \text{ dB}$$

**Electric and Magnetic Field Diffusion:**

$$E_i = E_o\, e^{-t/\delta} \qquad H_i = H_o\, e^{-t/\delta}$$

**Skin Depth:** $\delta = \frac{2.6}{\sqrt{f\,\mu_r \sigma_r}}$ in.




**Absorption Loss:**

$$A = 3.34\, t \sqrt{f\,\mu_r\,\sigma_r} \text{ dB}$$

**Where:**

- $E_O$, $E_i$ is the outside and inside electric field wave intensity
- $H_O$, $H_i$ is the respective magnetic fields
- t is the distance within the shield
- $\mu_r$ is the relative permeability (aluminum $\mu_r = 1$)
- $\sigma_r$ is the relative conductivity (aluminum $\sigma_r = 0.61$)

Figure 1

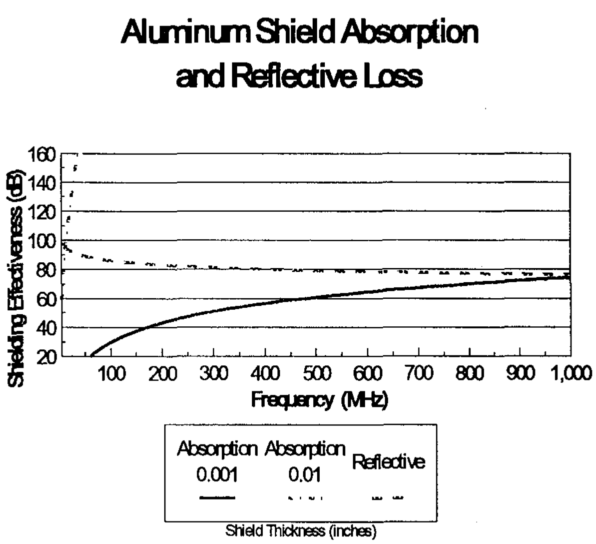
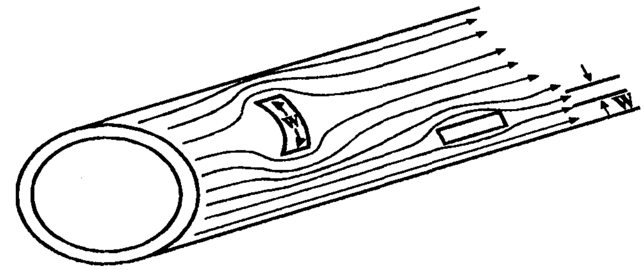



Figure 2

## Shielding Effectiveness As A Function of Aperture Leakage




$$S.E. = 20 \log\left(\frac{\lambda}{2W}\right) \text{ dB}$$

$$= 20 \log\left(\frac{5{,}906\ V_f}{fW}\right) \text{ dB}$$

**Where:**

- $V_f$ is the velocity factor (0.88 for foam T&D)
- f is the frequency in MHz
- W is the slot width in inches

Figure 3a

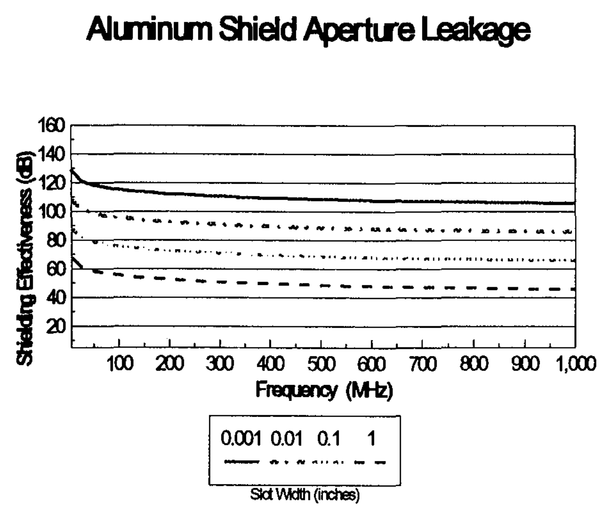



Figure 3b

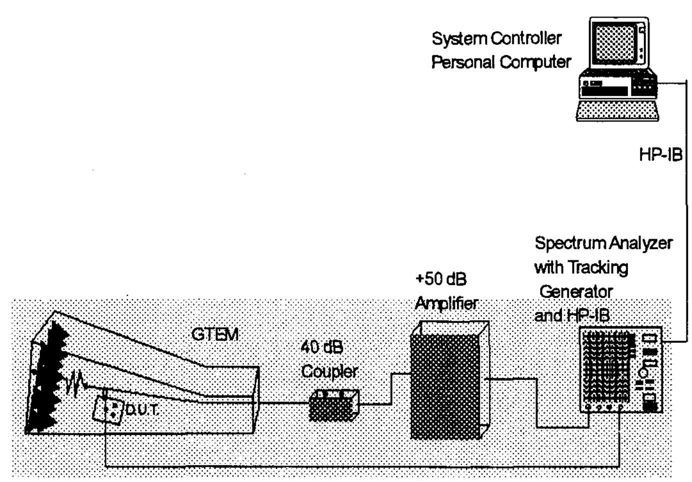



Figure 4

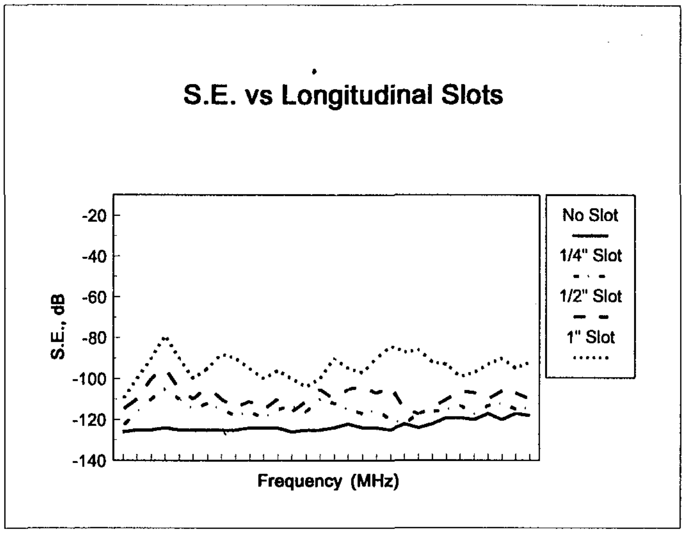



Figure 5a

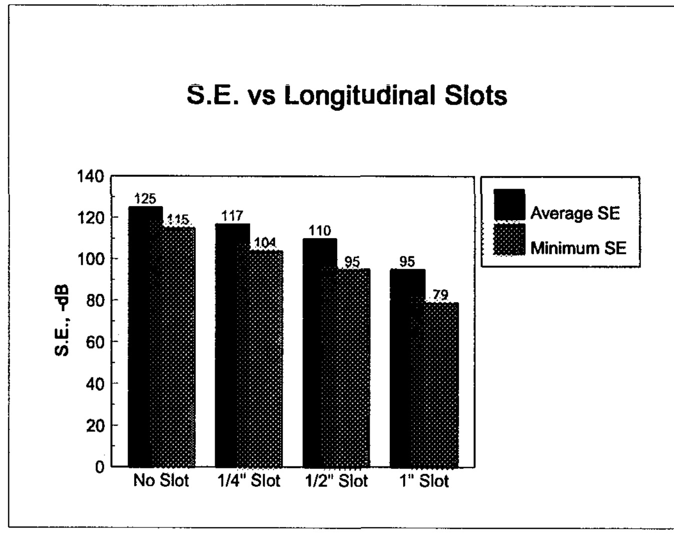



Figure 5b

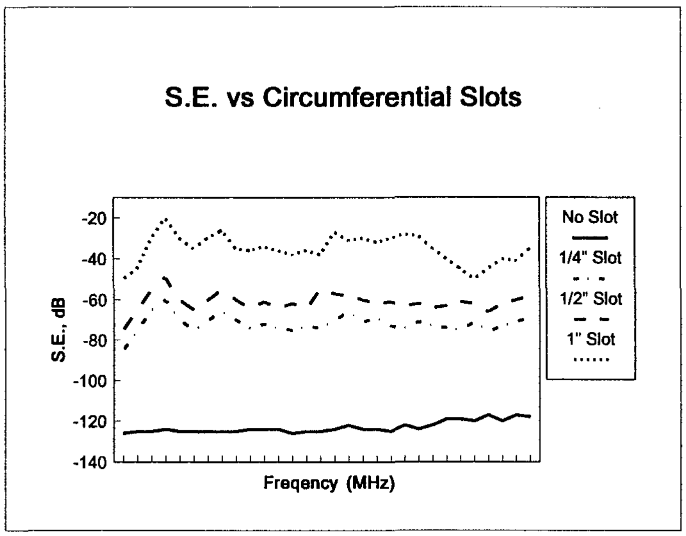



Figure 6a

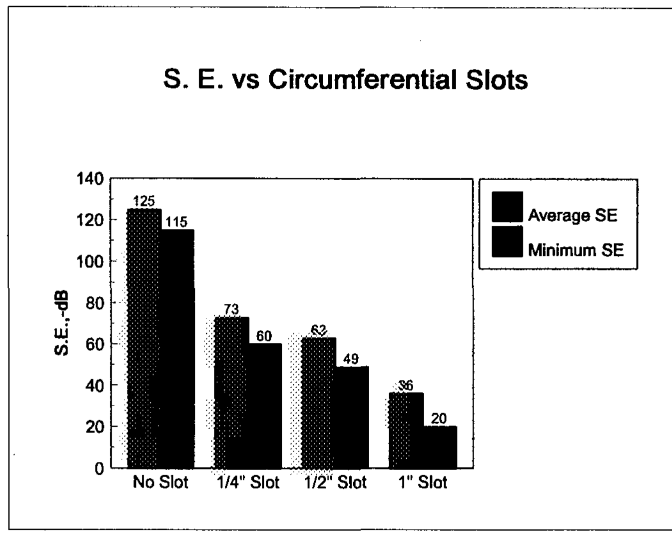



Figure 6b

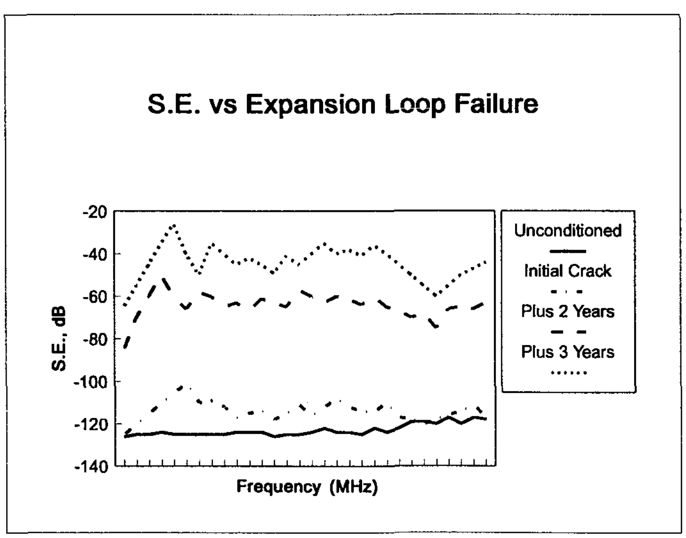



Figure 7a

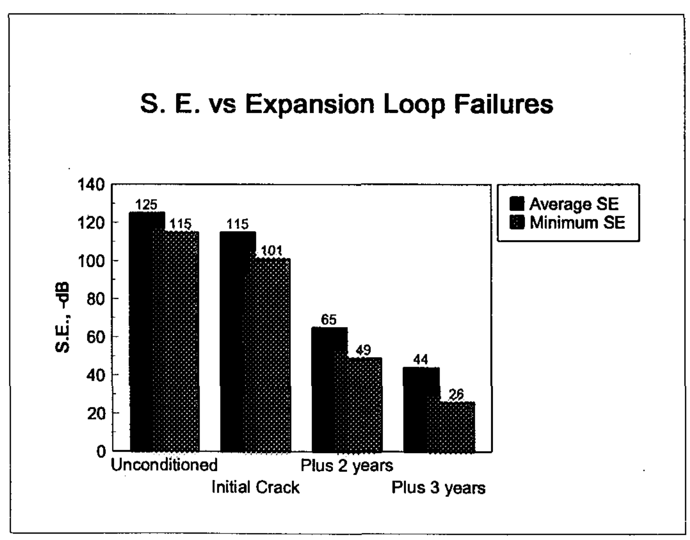



Figure 7b

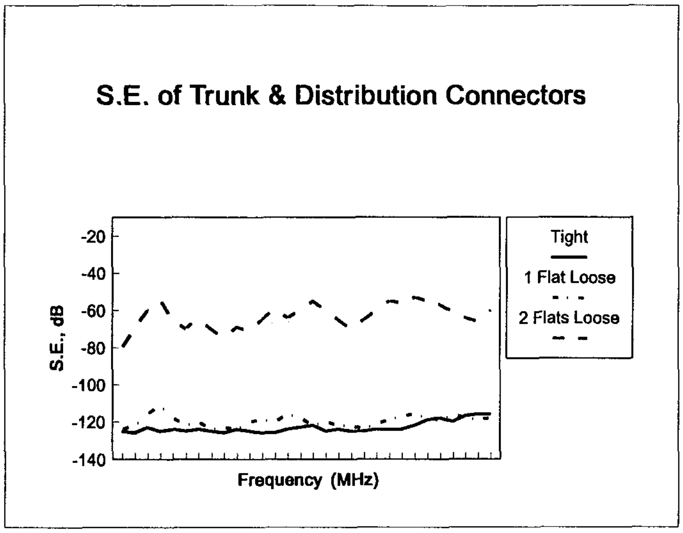



Figure 8a

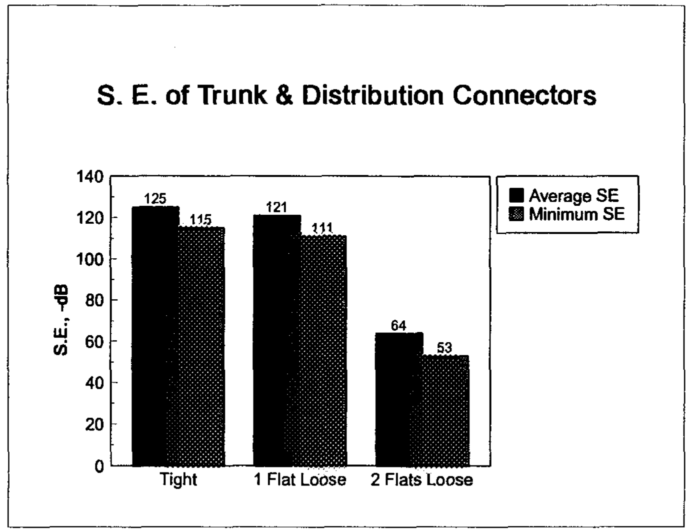



Figure 8b

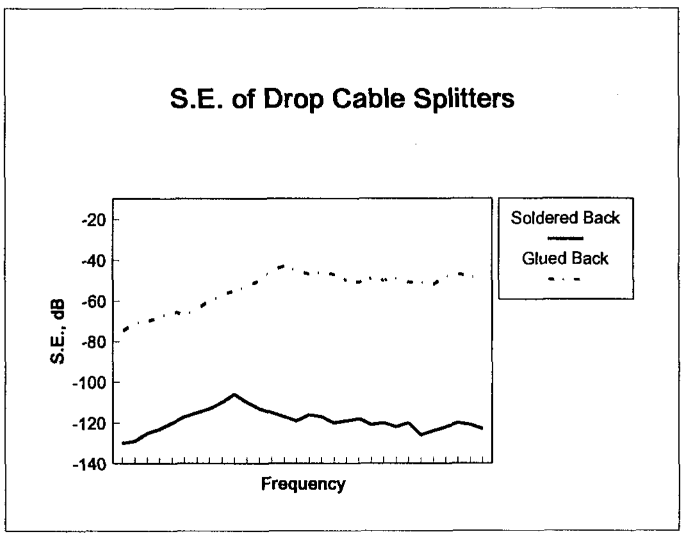



Figure 9a

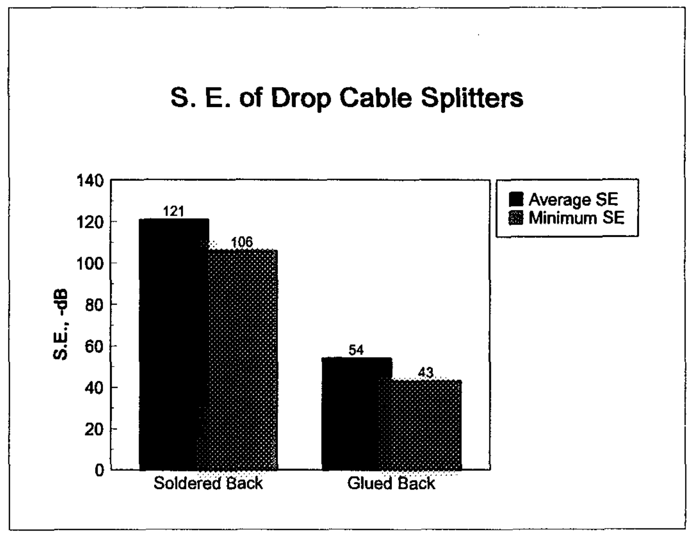



Figure 9b

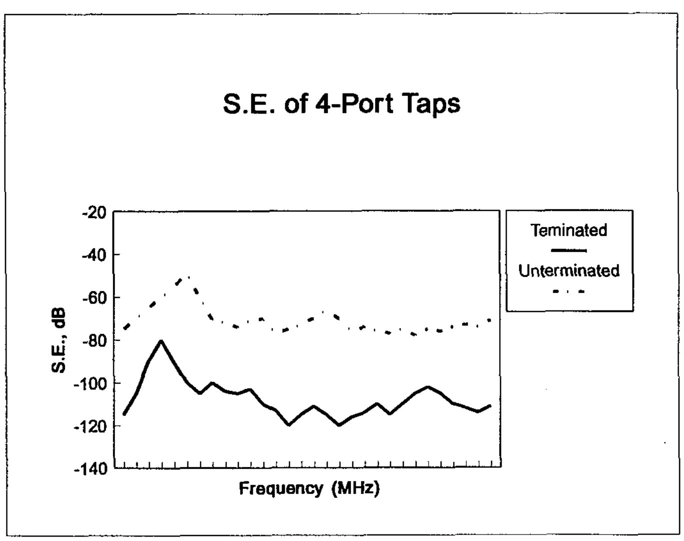



Figure 10a

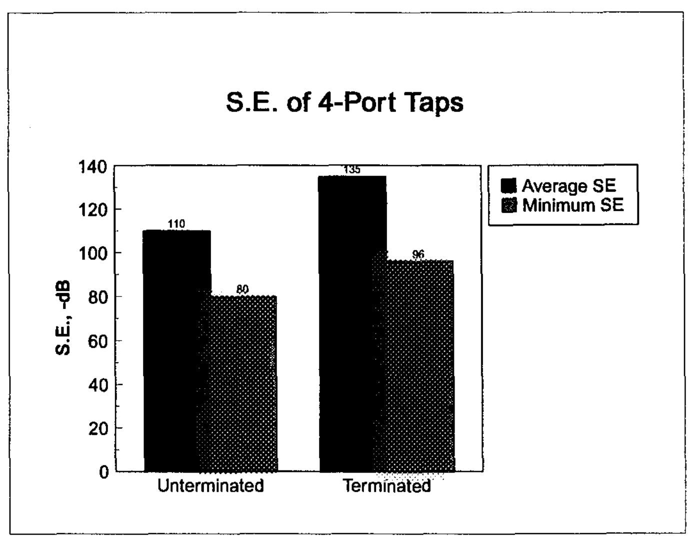



Figure 10b

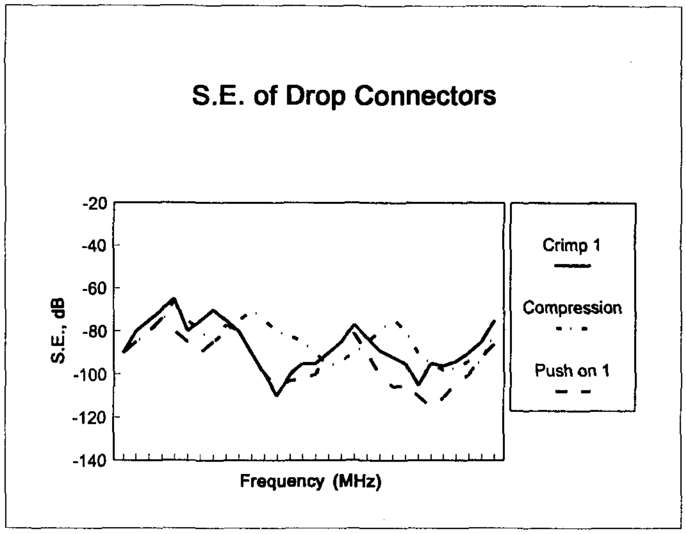



Figure 11a

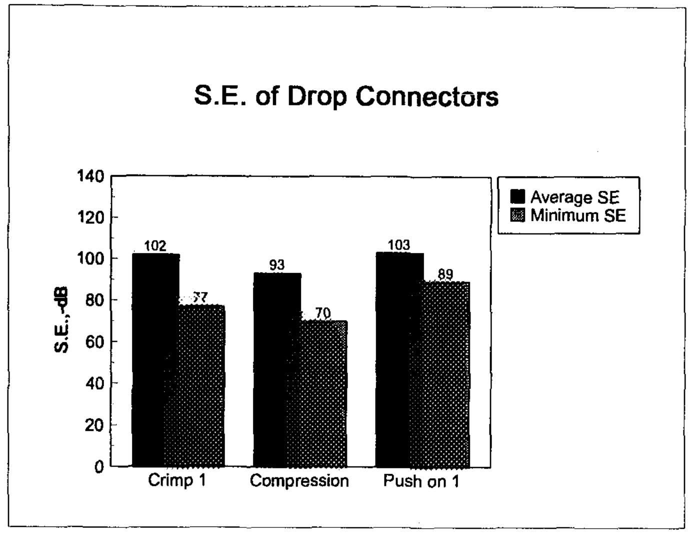



Figure 11b

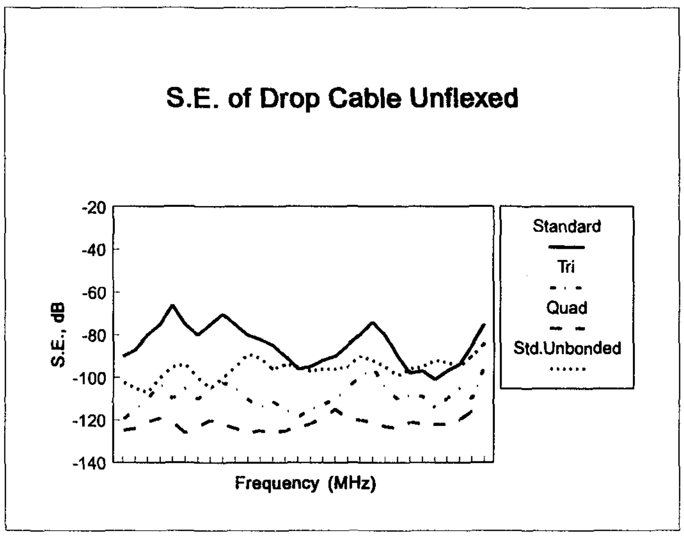



Figure 12a

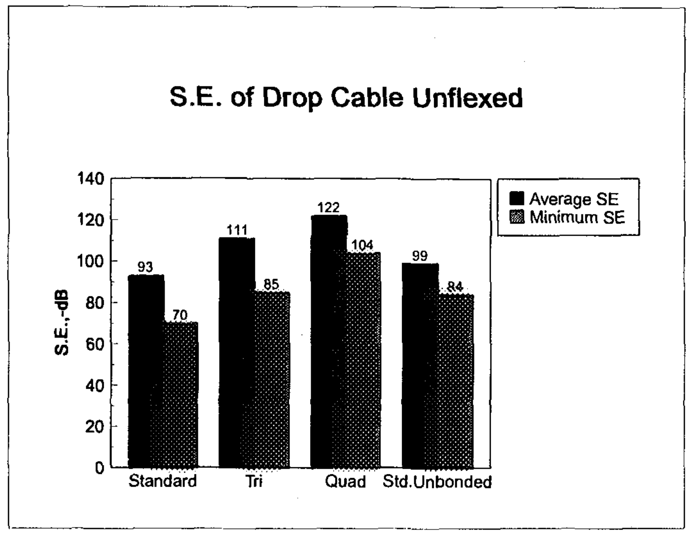



Figure 12b

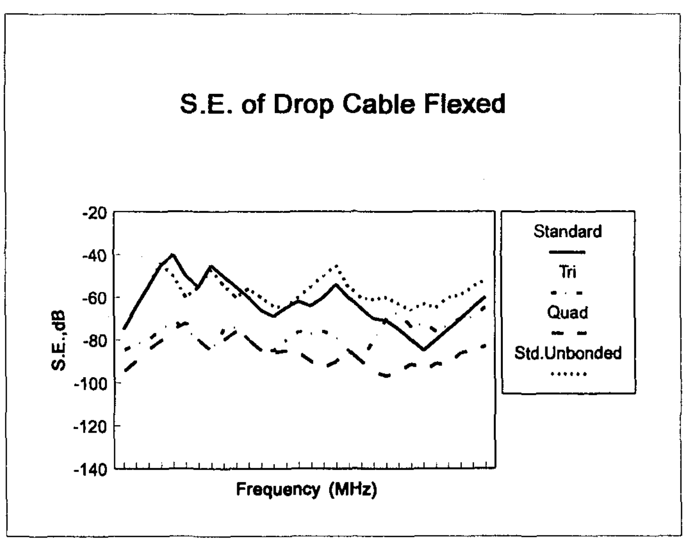



Figure 13a

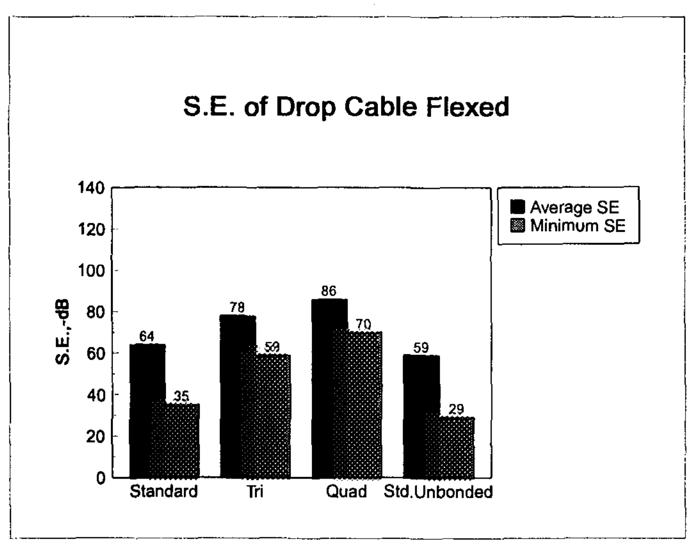



Figure 13b

# OPTIONS FOR CABLE COMPLIANCE WITH THE NEW EAS RULES

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ANTEC Corporation
*and*
Marty Callahan, President, HollyAnne Corp.

*Abstract*

*The 1992 Cable Act stipulated that cable systems would be required to participate in the new Emergency Broadcast System, but left details of compliance to the FCC. After much deliberation and consideration of comments by the NCTA, the SCTE EBS Subcommittee, and other interested parties, the FCC issued its Report and Order on December 9, 1994, amending Part 73, Subpart G of the Commission's Rules Regarding the Emergency Broadcast System.*

*This paper will examine those aspects of the Report and Order that pertain to cable television systems, and will present practical options for compliance under the new rules.*

## THE RULES

### Objectives

The FCC's overhaul of the current EBS system was undertaken with the express purpose of utilizing current technology to enable the President of the United States to exercise his/her communications-related powers in time of war or national emergency, while also allowing for more localized delivery of emergency messaging than the current EBS provides. Section 706 of the 1934 Communications Act grants the President the authority to take control of any or all stations within the FCC's jurisdiction under such circumstances. The new Emergency Alert System (EAS) is "designed to enable the President to exercise these powers quickly and efficiently."[1]

The Cable Act of 1992 amended the Communications Act by stating, "each cable operator shall comply with such standards as the Commission shall prescribe to ensure that viewers of video programming on cable systems are afforded the same emergency information as is afforded by the emergency broadcasting system..."

In the Report and Order, the Commission stated that the new rules were adopted with the Americans with Disabilities Act in mind, aiming to "make all facets of our society fully accessible to individuals with disabilities."

### Process

With the intent of serving the public interest, the FCC undertook a thorough examination of the existing EBS, and solicited comments from interested parties through a Notice of Inquiry and Notice of Proposed Rule Making (1991), and a Public Notice inviting comment on a 1992

NPRM/Further Notice of Proposed Rule Making. The FCC received 63 comments and 17 reply comments in response.

Field tests were conducted in 1993 to evaluate the availability and effectiveness of new technologies. The tests convinced the FCC that newer digital technology can provide far more flexibility and reliability than the old EBS system, and that manufacturers will be able to offer a wide range of necessary equipment for both broadcasters and cable systems. The field tests also confirmed that multiple types of transmission systems could relay the digital information, and are in fact, complementary.

The FCC also conducted numerous meetings and regional workshops with state and local emergency officials, manufacturers, and interested parties from the telecommunications industry, to explore the technical and operational issues surrounding implementation of a new EBS. The Society of Cable Television Engineers' EBS Subcommittee and the NCTA were particularly active in these proceedings, and were instrumental in the final rules' determination that allows cable operators to select one of two basic options for compliance, based on cost effectiveness.

Deadlines

By July 1, 1997, cable systems must have installed and operating all necessary EAS equipment to receive, decode and encode the new digital emergency information, and to transmit the required information across all cable channels. Broadcast stations must meet a compliance deadline of July 1, 1996.

Functional Mandates

The FCC states in paragraphs 58 and 59 of the Report and Order, "We...require cable systems to maintain EAS equipment in accordance with the rules we adopt in this order. For national emergencies, we are requiring compulsory retransmission of Presidential EAS messages. Cable systems must interrupt all channels and provide information to subscribers. Although not required, we also encourage EAS activation for state or local emergencies.[2]

"Cable systems must transmit the national EAS message codes, the attention signal, emergency message, and end of message (EOM) code to their subscribers...The equipment must provide an audio message on all channels for those event codes selected by cable company personnel. Cable systems may elect not to interrupt EAS messages from broadcast stations upon written agreement among all concerned."

The FCC outlines two basic alternatives for complying with the mandate, presented here as Option 1 and Option 2. Refer to Figure 1 and Table 1 for a headend block diagram and sample bill of materials, respectively, for Option 1. Option 2 block diagram and sample bill of materials are presented in Figure 2 and Table 2.

Option 1 Equipment Required

This option provides both audio and video information (text appearing on-screen at a location that does not interfere with closed-caption messaging) across all channels on the cable system. Deaf subscribers are alerted by the video information on every channel, and the blind are alerted by audio on every channel, thus satisfying the requirements of the Americans with Disabilities Act. The FCC does not require cable systems to alert subscribers whose television sets are not turned on.

Option 1 requires that the EAS receiver/decoder drive a character generator, and combine both outputs in an I.F. controller/modulator. The I.F. controller in turn feeds an I.F. signal carrying the audio and video messages into each modulator and processor in the headend. When an emergency or test message is received, the I.F. controller activates the I.F. switching modules on each modulator/processor, and overrides the regular programming until the End of Message code is received. Normal programming is then resumed. A printer is also required, to log all emergency and test messages. (See Figure 1 and Table 1)

Cable systems with IF switching already installed for every channel (e.g., Scientific-Atlanta models 6340, 9270, or 9280) may find this option cost-effective. While the installation requires direct input into each channel's I.F. port and is therefore somewhat labor-intensive, larger cable systems serving many deaf subscribers may realize cost savings over Option 2.

Total cost to upgrade an I.F.-ready headend is estimated at $5900-$8700. Additional costs to add I.F. switching capabilities may run $130 to $155 per channel, depending on manufacturer and model.

Option 2 Equipment Required

Option 2 provides audio override with video interrupt (displacement of the picture with black, blank or flashing screen for short periods) across all channels, with audio and video override on one channel. Under Option 2, an equivalent alerting function must be provided to deaf and hard-of-hearing subscribers throughout the system. This requirement can be satisfied through separate in-home devices that activate strobe lights, bed-shakers, etc., and may also carry audio alerts, voice warnings, and visual text messages.[3]

Equipment necessary to provide Option 2 functionality is somewhat simpler, since it is installed after the combining network, and does not require separate inputs to each processor and modulator on the system. (See Figure 2 and Table 2) The EAS receiver/decoder drives a character generator whose output feeds a modulator for the designated audio/video channel. Combined with an audio input from the EAS decoder, the modulator's output feeds a comb generator that also takes inputs from the combining network, and control and audio inputs from the EAS decoder. The R.F. signal from the comb generator feeds the distribution system. A separate, 52Mhz carrier from the EAS decoder delivers the same emergency messaging to activate in-home devices for the deaf. A printer is also required in Option 2, to log emergency and test messages.

Option 2 will appeal to cable operators whose headends are not already equipped with I.F. input and switching modules, and to those who estimate a relatively small number of deaf subscribers on their systems. In-home devices to support the deaf community may cost \$65-\$95 each, depending on final determinations by the FCC on what features will be required (e.g., simple contact closures to activate strobe, vs. LCD readout to deliver complete emergency message on the device). Headend costs for Option 2 are estimated at \$5685-\$9840.

## The EAS Receiver/Decoder

Several manufacturers (including TFT and Idea/onics) are developing equipment solutions for EAS compliance. For simplicity's sake, this paper presents the MIP-921 (Multiple Input Processor) developed by HollyAnne Corp., as it combines in one unit many functions that must otherwise be accomplished in several pieces of equipment. Any EAS receiver/decoder must perform the following functions, by FCC requirement:

1. Capable of using the EAS protocol to send and receive EAS alerts.
2. Certified in accordance with Part 2 of the Commission's Rules, 47 C.F.R. Part 2.
3. Automatic override for national (Presidential) messages
4. Means to decode EAS messages either automatically or manually
5. Audio inputs for two EAS monitoring assignments, and one data input to receive another transmission mode
6. Means to store at least two minutes of audio or text messages and at least 10 preselected message codes
7. Ability to display any valid EAS message codes to show the originator, event, location, valid time period of the message and the local time the message was transmitted. Capability selectively to receive originator, event and location codes
8. Security measures to ensure that there will be no unauthorized personnel changing preselected codes
9. Programmable so that operators can select certain message codes in advance; cable management will determine which message codes in

their decoder will automatically interrupt programming for emergencies affecting their audience, including setting priorities for state and local area emergencies. All decoders must be programmed for EAS national level emergencies and required tests.

10. Must allow for relaying EAS alerts to narrowly targeted audiences; must have a display that will show who originated the message, the event or reason for the message, location of the event, and the valid time period of the event in local time.[4]

Test and Operation Requirements

The FCC requires that weekly tests of the EAS system be conducted, although three tests per month may be "unobtrusive", or silent tests, with no interruption of regular programming.

However, one test per month must be conducted on-air, and these tests will involve the following:

1. Transmission of the digital codes, transmission of at least 8 seconds of the attention signal, test audio script and EOM code.
2. All AM, FM, and TV stations and cable systems within the prescribed test area, as defined by the State Emergency Communications Committee (SECC), must transmit the test within the same 15 minute period.
3. Unattended broadcast stations and cable systems must have a record available that shows EAS test and alert transmissions.
4. Tests in odd numbered months should occur between 8:30 am and local sunset. Tests in even numbered months should occur at times other than between 8:30 am and sunset.
5. Tests will originate from key EAS sources identified in the state plan.
6. The test schedule and script content will be developed by the SECCs and the Local Emergency Communications Committees (LECCs) in cooperation with affected stations and cable systems, the National Weather Service, and emergency management officials.[5]

It is clearly in the best interest of cable operators to become directly involved in local and state committees, as the scheduling of disruptive tests will otherwise be determined without consideration for the concerns of cable operations.

The FCC "expect(s) these committees to reorganize to include cable systems ...to hold elections for positions, to organize training, and to approve the policies and procedures for EAS operations, tests and activations within their domain. We anticipate that each SECC will have a Broadcast co-chair and Cable co-chair to ensure that their industries are fully participating in the development of state emergency plans."[6]

Unclear and Undetermined Issues

While the FCC was very thorough in its deliberation process and explication of the new EAS requirements in the Report and Order, a number of issues are still subject to interpretation; still others are unresolved at this writing, awaiting comments and reply comments on the Further Notice of Proposed Rule Making.

One of the issues cable operators will seek clarification on, is the provision of in-home alerting devices for deaf subscribers. Is a cable operator obligated to provide such devices to homes where only one of several residents is deaf or hard of hearing (a deaf child of hearing parents, for example)? What is the minimum level of functionality required for such a device?

The Further Notice of Proposed Rule Making asked for comments on the economic impact of EAS compliance on small cable systems, and the possibility of waivers or exemptions for those systems, and the size requirements for systems to qualify for such treatment.

The FNPR also seeks comment on whether the FCC should preempt application of conflicting requirements of local franchising authorities, should they pose any threat to the federal objective of maintaining EAS nationwide alerting capability.[7]

While the FCC mandated participation in the new EAS by broadcasting and cable, they have encouraged the voluntary participation of video dialtone providers, DBS, MMDS, telephone and cellular carriers, and other service providers. The FNPR also seeks comments on whether any or all of these services should be required to participate, as well.

Conclusion

While cable compliance with the new EAS may generate economic hardships for very small cable systems, the equipment required to satisfy the FCC rules is not terribly cumbersome, and will generally be available for well under $10,000 per head end. The new EAS undeniably adds the benefits of reliably and automatically alerting the public of emergencies, and does indeed satisfy the national objective of serving the public interest. Cable operators who can embrace implementation of this new system will have the opportunity to tell an important story in their communities about providing a service with the potential to save lives and property.

*Acknowledgments*

*The authors wish to thank Ken Wright and Steve Johnson of the SCTE EBS Subcommittee, and Wendell Bailey of the NCTA for their important work with the FCC, and for their success in reducing the burden of compliance for cable operators. The authors also thank John*

*Grothendick and Michele Dionne of ANTEC, and Ken Cannon of Scientific-Atlanta, for their contributions to this paper.*

[1] FCC 94-288, Report and Order and Further Notice of Proposed Rule Making, paragraph 5
[2] Where franchise agreements dictate local alerting, the national alerting requirements of the FCC do not supersede franchise authorities; local alerting must be continued in accordance with franchise agreements.
[3] id., paragraphs 61-62; also footnote 69, re. currently available devices in comments of Safety Alert Monitor, November 12, 1993.
[4] id., paragraphs 87 through 94.
[5] id., paragraphs 107 through 112.
[6] id., paragraph 132.
[7] id., paragraph 153, Section A of Further Notice of Proposed Rule Making.

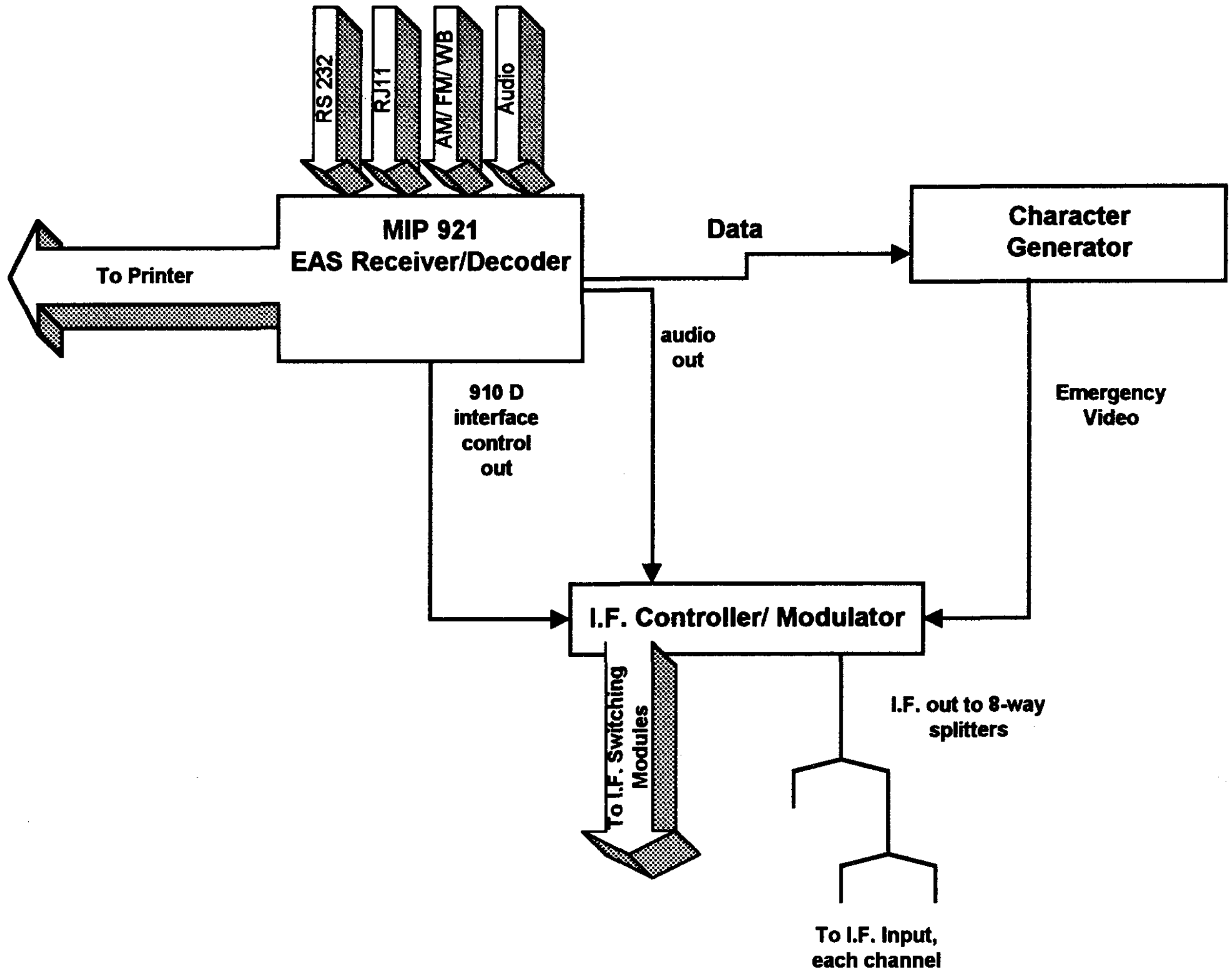



**Figure 1**

**Option 1: Audio/Video Override on All Channels**

## OPTION 1: EAS EQUIPMENT REQUIREMENTS

### Audio/Video override on all channels

| Equipment Description | Quantity | Approx. Price |
|---|---|---|
| EAS Multiple Input Processor, receiver and decoder, with AM/FM and WB receivers | 1 | $1500- $4500* |
| Event/Status Printer | 1 | $150-$300 |
| Character Generator | 1 | $700-$800 |
| I.F. Controller/Modulator | 1 | $3500-$3800 |
| High Level I.F. Switching Option ** | 1/channel | $130-$155 per channel |
| Highly-shielded 8-way splitters, in series | 1/ 7 channels | $50- $100 |
| Total (I.F.-ready headend) | | $5900- $9500 |

* High end units may include character generator

** I.F. input switch must be installed in modulators and processors to deploy Option 1. Some models may be upgraded with external switches, but consult with headend equipment manufacturer for details.

**Table 1**

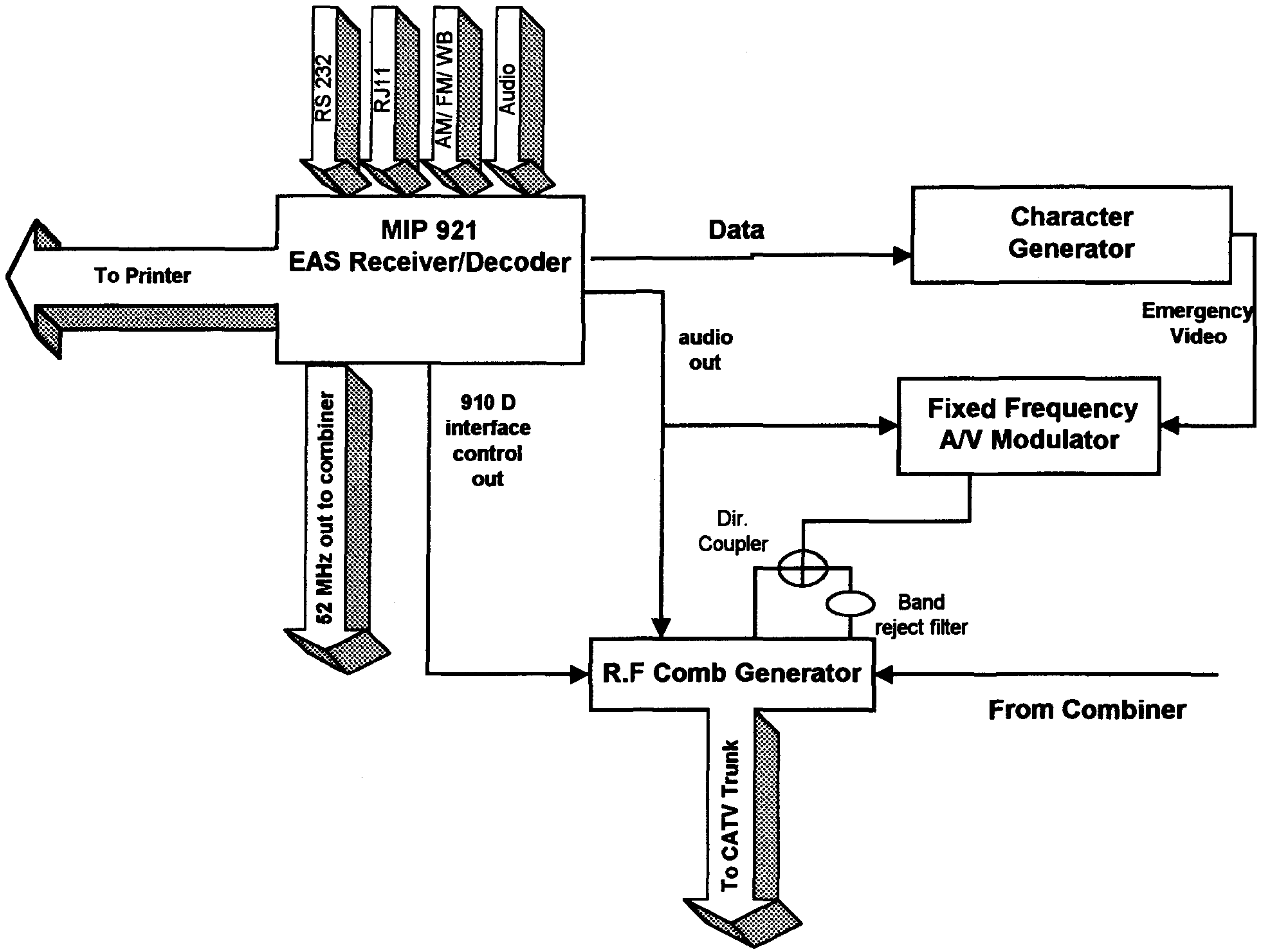



**Figure 2**

**Option 2: All Channel Audio Override with Video Interrupt**

## OPTION 2: EAS EQUIPMENT REQUIREMENTS

### All channel audio override with video interrupt

| Equipment Description | Quantity | Approx. Price |
|---|---|---|
| EAS Multiple Input Processor, receiver and decoder | 1 | $1500- $4500* |
| Event/Status Printer | 1 | $150-$300 |
| Character Generator | 1 | $700-$800 |
| Low-cost fixed frequency audio/video modulator | 1 | $135-$340 |
| R.F. FM audio comb generator | 1 | $3200-$4700 |
| In-home alerting device for deaf subscribers | 1/ deaf home | $65-$95 each |
| Total headend costs (excluding in-home devices) | | $5685- $9840 |

* High end units may include character generator

**Table 2**

# Performance Results of a Low-cost Alternative Equalizer Architecture for 64/256-QAM Demodulation in a CATV Receiver

**K. Laudel**
**Applied Signal Technology, Inc.**

## ABSTRACT

*The viability of high order (64,256) QAM transmission techniques in a cable environment have recently been technically proven in extensive testing [1,2,3]. However, the economic goals of cable TV set-top terminal solutions for the demodulation of 64- and 256-QAM remain as the engineering challenge for modem designers. Current pricing for QAM demodulator chips will require a fairly significant reduction in cost within the next year to meet consumer demands as well as the competitive demands from market forces such as Direct Broadcast Satellite.*

*One expedient method for cost reduction exists in the demodulation algorithms. Within the demodulator architecture, it is not uncommon for the adaptive equalizer to occupy upwards of 40–50 percent of the silicon area. Traditional, T-spaced and T/2-spaced equalizers have been the cornerstone of many of the currently proposed architectures [1,4]. However, there exists a strong economic motivation in the consumer marketplace for examining alternative equalizer structures in order to decrease the silicon area. The intent of this paper is to address the technical performance capabilities of an alternative equalizer architecture that could potentially result in an equalizer die area savings of approximately 33 percent. More specifically, laboratory and field test results of a T/2-spaced prototype will be discussed and compared to simulation results of the proposed design.*

## INTRODUCTION

The advent of low-cost digital compression techniques for use in the CATV industry ushered in the search for a means of a high capacity method for digital transmission. Two likely candidates, QAM (Quadrature Amplitude Modulation) and VSB (Vestigial Side Band), have been tested in prototype equipment in both simulated and live cable systems **[1,2,3,5,6]**. Both methods are efficient transmission techniques and are equivalent in terms of the number of bits/Hz that can be transmitted in a 6 MHz channel. For the purposes of this paper, the discussion will be limited to QAM systems, although extensive literature on VSB exists **[4,5,6,7]**. QAM has become the logical transmission choice for several key set-top terminal manufacturers because of its widespread success in digital microwave radios and voiceband modems, as well as its adoption as the standard in Europe for Digital Video Broadcasting (DVB).

The generation of a QAM signal is theoretically straightforward. The information bit stream is demultiplexed into in-phase and quadrature rails. Each rail encodes its bit stream into $2^n$ levels and then bandlimits the signal with baseband filters in order to limit the resultant signal to a 6 MHz band. The filtered baseband signals are then multiplied by two quadrature tones, which are typically at TV IF (43.75 MHz). The resultants are then summed together to produce a 6 MHz-wide signal centered at TV IF. In a typical Head End, this signal is then upconverted to the destination channel frequency and transmitted via the cable. The function of QAM demodulation generally is found in the set-top terminal block diagram between the A/D Converter and Forward Error Correction as depicted in **Figure 1**. Its purpose is generally the following:

1. Extraction of the in-phase (I) and quadrature (Q) rails.*
2. Baud or symbol timing recovery.
3. Carrier recovery.
4. Automatic Gain Control (AGC).
5. Channel equalization.
6. Provide symbol decisions for the Forward Error Correction.

* Functions which are also commonly performed in analog circuitry

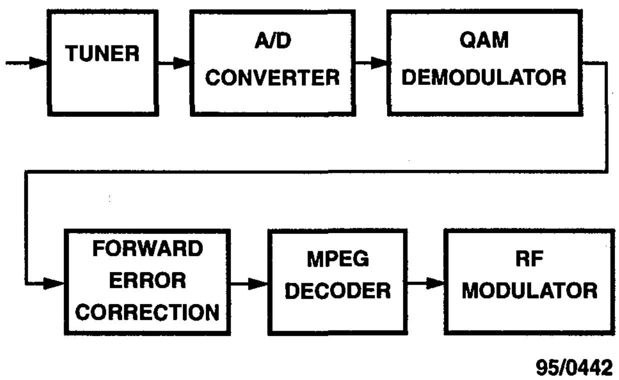
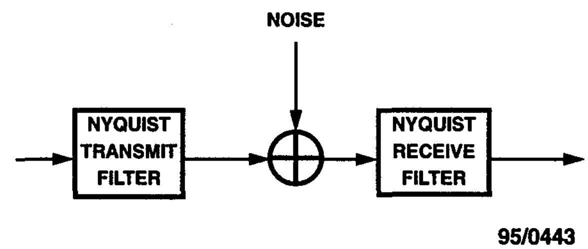



**Figure 1. Set-top Terminal Signal Processing Chain**

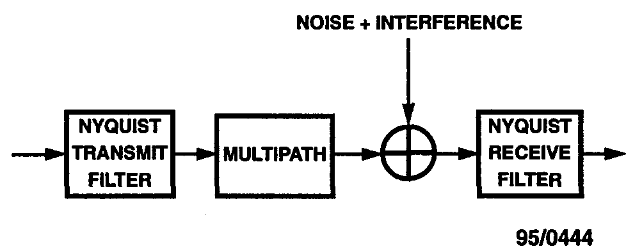
Channel equalization is necessary to mitigate the effects of echoes caused by impedance mismatches, co-channel interferers, adjacent channel interferers, and amplitude distortions caused by other set-top terminal components such as SAW filters. The function of conventional channel equalization can typically consume 40–50 percent of the silicon area in VLSI implementations of a QAM demodulator. While it is true that advances in silicon fabrication processes (i.e., 0.7 micron to 0.35 micron) and full custom layouts will eventually drive the cost of silicon down, more expedient algorithmic methods are needed in the short term in order to meet the price goals set by consumers as well as other market forces such as Direct Broadcast Satellite. Therefore, there exists significant motivation towards examining alternative equalizer architectures.

## Traditional Approaches

A simplistic view of a digital communication link is shown in **Figures 2(a) and (b)**. In **Figure 2(a)** the idealized case of a channel consisting of additive white Gaussian noise is shown. In this case a Nyquist transmit filter is employed and the requirements at the receiver are relatively straightforward. Since the goal at the receiver is to minimize the intersymbol interference (ISI), a simple matched Nyquist filter at the receive site will suffice. However, in real cable systems the link model contains additional interference phenomena such as microreflections caused by mismatches in impedances and co-channel interferers caused by NTSC carriers. Thus, a more realistic model for the link is shown in **Figure 2(b)**. The channel model for the multipath can be described as adding a pole to the overall channel response. In this case a simple Nyquist filter at the receiver will not be capable of removing the resultant ISI since the symbol nulls have been corrupted and the result is a smearing of energy from one adjacent symbol to the next, which can result in incorrect symbol decisions.




**(a) Simplified Digital Communication Link**




**(b) Digital Communication Link with Co-channel Interferers and Multipath**

**Figure 2. Digital Communication Links**

The poles created by the microreflection model require inverse filtering, i.e., an inverse filter with a zero. Since these microreflections can occur at variable amplitudes and delays, an adaptive equalizer is typically used. **Figures 3(a) and (b)** show two common feedforward linear Finite Impulse Response (FIR) equalizer architectures. In **Figure 3(a)** a T-spaced architecture is shown. "T-spaced" refers to the spacing or delay between adjacent taps, which in this case is one baud or symbol period. **Figure 3(b)** contains a T/2-spaced architecture, which simply means that the spacing between taps is one-half of a baud period. Many of the published, proposed designs have employed T- or T/2-spaced architectures **[1,4]**.

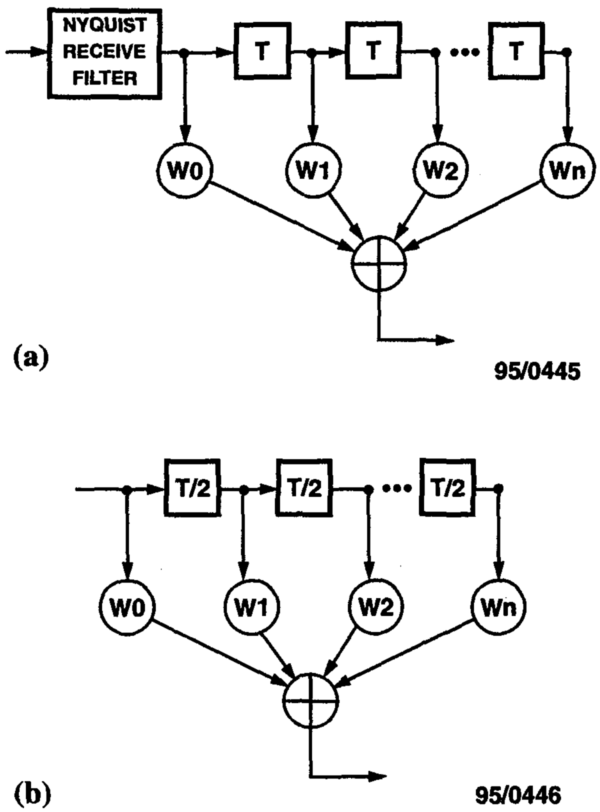



**Figure 3. (a) T-Spaced and (b) T/2-Spaced Equalizers**

There are several issues involved in choosing one spacing over another. One advantage of the T/2 architecture is that it is capable of aiding symbol timing recovery in attaining optimal baud phase. In addition, a separate Nyquist receive filter is needed before the T-spaced equalizer since this structure is not capable of synthesizing the matched receive filter. Thus, a T/2 architecture was chosen by Applied Signal Technology for the development of a prototype system for use in characterizing performance in both "live" and simulated cable environments. This prototype system would also serve as the design basis for a VLSI architecture for use in set-top terminal QAM receivers. The T/2 prototype has been through extensive testing at the Advanced Television Test Center (ATTC), in the laboratory, and in "live" cable systems using existing NTSC gear and analog set-top tuners. Results of this testing will be examined and referenced throughout this paper as a point of comparison with the proposed alternative architecture.

## PERFORMANCE RESULTS

### Performance Results of a T/2-Spaced Demodulator

*Results of the ATTC/CableLabs 64/256-QAM Test*

Two rounds of extensive testing were performed on the Applied Signal Technology T/2-spaced prototype at the Advanced Television Test Center (ATTC). The first round of tests was conducted in January by CableLabs, Inc., as part of the evaluation process for proposed HDTV transmission systems. In this round of testing the prototype was tested solely in the 256-QAM mode of operation. The second round of testing involved 64-QAM only and was conducted by CableLabs in October. For both rounds of testing impairment scenarios were developed in order to fully characterize each proponent's system and determine robustness. Thus, impairments such as microreflections, Gaussian noise, phase noise, residual FM, carrier offset, CTB and CSO, AM hum, etc. were added to the test signal for full system characterization and determination of each system's point of failure. The results are shown in **Table 1**. Results which are largely dependent upon the equalizer are shown with an asterisk (*).

The "Channel Acquisition Time" is germane to this discussion since this test consisted of switching from an unimpaired channel to a channel (12) with impairments which included microreflections. Case 1 consisted of a 300 nsec, –18 dB ghost on channel 12. Case 2 consisted of Gaussian noise at a level which is 3 dB below the threshold of visibility (TOV—defined as the coded Bit Error Rate (BER) of 3.0E–06), and a long delay ray of 2.5 µsec which was –20 dB down from the signal. Case 4 included both Gaussian noise (–3 dB below TOV) and a microreflection which was –20 dB down at 600 nsec delay. Note that the T/2-prototype was capable of acquiring within 0.5 seconds for both 64- and 256-QAM, which is satisfactory for the majority of consumers. Since the 256-QAM testing in January, the acquisition times for the Applied Signal Technology prototype were improved to be on the order of 250 msec, which is extremely quick even by today's NTSC standards. Note that the 64-QAM acquisition times reflect this improvement, cutting down the acquisition times by a factor of two. This currently holds true for 256-QAM as well.

## Table 1: 64/256-QAM Test Results, Cable Labs

| | 64-QAM | 256-QAM |
|---|---|---|
| **C/N (dB)** (1) * | 21.2 | 29 |
| **CSO (dBc)** (1) * | –27.95 | –38 |
| **CTB (dBc)** (1) * | –41 | –48 |
| **Phase Noise (dBc/Hz)** (1,2) | –78.22 | –85 |
| **Residual FM (kHz)** (1,3) | >99 | 66 |
| **Hum (% Modulation)** (1,4) | 13.8 | 5.7 |
| **Tuner Pull-in Range (kHz)** (5) | –165/+300 | ±300 |
| **Channel Acquisition Time (Sec.)** * | | |
| No impairments | 0.234 | 0.48 |
| 300 nsec, –18 dB ghost on test channel | 0.291 | 0.52 |
| 2.5 µsec, –20 dB ghost plus noise at –3 dB from TOV on test channel | 0.284 | 0.54 |
| Noise at –3 dB from TOV plus phase noise at –3 dB from TOV on test channel | 0.263 | 0.64 |
| 600 nsec, –20 dB ghost plus noise at –3 dB from TOV on test channel | 0.404 | 0.55 |
| Average | 0.309 | 0.55 |
| **Isolation Between Receivers** * | | |
| Echo (dB) at 0 BER | –20.9 | –24 |
| Data loss/hit at +6 dB Echo (sec.) | 0.22 | 0.046 |
| **CW Interference** * | | |
| C/I (dB) Continuous (208 MHz) | 23.65 | |
| C/I (dB) gated (205.25 MHz) | 21.65 | |
| **Adjacent Channel Interference** * | | |
| Degradation in threshold C/N caused by equal power adjacent QAM signal (dB) | 0.25 | |
| Degradation in threshold C/N caused by adjacent NTSC @ +7.64 dB (dB) | 0.25 | |
| **Grand Alliance Test** * | | |
| BER | Not Tested | Passed |
| Acquisition | Not Tested | Passed |

(1) TOV <3.0E–06 after error correction.
(2) Measured at 20 kHz offset
(3) Peak values. Carrier modulated with quasi-rectangular 120 Hz signal.
(4) Carrier Am modulated at 120 Hz.
(5) Threshold in <5 sec for acquisition. No impairments.

Another test based on a real life cable impairment scenario requiring robust equalization is the "Isolation Between Receivers" test. This test simulates a remote viewer channel surfing on another receiver connected to the same splitter as the demodulator under test. The test varies the level of a 150 nsec echo which is being pulsed off by virtue of a switch for 40 msec at a 5-second repetition rate. The amplitude of the echo is increased until errors occur ("Echo at 0 BER"). Note that the prototype was error free for an isolation of 24 dB, which can be achievable in a consumer grade splitter for reasonably low cost.

Finally, a multiple impairment test was devised by the Grand Alliance for which the BER was checked to ensure it was below TOV and the acquisition time was less than or equal to 0.5 seconds. The impairments consisted of 33.5 dB C/N, 3% AM hum, and an echo –13 dB down at a delay of 600 nsec. The T/2 prototype performed well and exceeded the specification for 256-QAM.

Additional testing was done on the 64-QAM system relative to microreflections as single impairments (see **Table 2**). Acquisition threshold and TOV tests were run with echo delays ranging from 0.5 to 5 microseconds. Performance proved to be sufficiently robust for real cable systems. It has been shown in one particular study involving measurements taken at approximately three hundred subscriber homes in twenty cable systems that 99 percent of all microreflections measured had echo amplitudes less than –19 dB below the desired signal and delay times less than 1.28 microseconds **[8]**. After the testing at CableLabs was completed, an independent test was performed on the demodulator only (without the tuner) at Applied Signal Technology in order to determine 256-QAM performance in the presence of microreflections as single impairments. The results are shown in **Table 2**. Results are within 1.5 dB of the 64-QAM performance for the shorter echo at 0.5 microseconds, but start to diverge slightly as the echoes get longer. This makes sense since the equalizer need only synthesize a single notch for the shorter echoes in order

**Table 2: 64-QAM Echo Test Results**

| | **64-QAM Tested by CableLabs (1)** | | **256-QAM Tested by Applied Signal Technology (2)** |
|---|---|---|---|
| **Delay (μs)** | **Acquisition (dB relative to signal** | **Threshold (TOV) (dB relative to signal)** | **Threshold (TOV) (dB relative to signal)** |
| 0.5 | –5 | –3 | –4.5 |
| 1 | –6 | –4 | |
| 1.5 | –6 | –5 | –11.5 |
| 2 | –7 | –7 | –12.5 |
| 2.5 | –10 | –10 | |
| 3 | –11 | –10.5 | |
| 3.5 | –11 | –11 | |
| 4 | –11 | –10.5 | |
| 4.5 | –10.5 | –10.5 | |
| 5 | –20 | –20 | |

(1) 64 taps, initialized at tap 16

(2) 32 taps, initialized at tap 8

to mitigate the distortion. When the echoes get longer, the amplitude ripple or "scalloping" in the frequency domain occurs with increasing frequency. The equalizer has a harder time synthesizing an increasing number of notches for a given number of taps and this, combined with the additional sensitivity inherent in 256-QAM, causes the aforementioned diversion of performance for the longer echoes. As can be seen from **Table 2**, 256-QAM performance for TOV (29 dB C/N) is approximately 6.5 dB from 64-QAM performance for a ~1.5 microsecond echo. However, this performance is also sufficient for the majority of cable systems.

### *Results of the 256-QAM New York City Field Test*

Between February 24 and March 3, 1994, a three-phase test was conducted by OmniBox, Inc., a developer of interactive television networks, over the New York City cable system of Time Warner Cable. All hardware used during the testing was standard NTSC equipment normally employed in the system, except that which provided the 256-QAM input signal to the upconverter, and the QAM demodulator. The test equipment did not provide any Forward Error Correction (FEC) or interleaving, and all BER numbers were raw, uncoded numbers. The test was set up in three phases, with each phase designed to examine different aspects of television transmission and reception through a typical cable system, thereby identifying degradation associated with different system elements. Phase One consisted of sending the 256-QAM signal across a 22-kilometer fiber optic link on channel 41 between the Head Ends at Brooklyn Queens Division (Flushing, Queens) and Manhattan Cable (23rd Street). Phase Two consisted of transmitting the QAM signal over standard coax cable and through the second longest cascade of trunk amplifiers in New York City with performance being measured at the end of the distribution chain at the southern tip of Manhattan. Phase Three was set up to receive the 256-QAM signal at a subscriber site (within Manhattan Cable's building) through approximately four trunk amplifiers and one bridge amplifier plus numerous splits and taps. These three phases of testing were selected as the best way to characterize the transmission and reception of 256-QAM over a typical cable system.

The Phase One test setup is shown in **Figure 4**. In this phase, the QAM signal was carried on channel 41 and the entire fiber optic line was terminated at the Manhattan Cable Head End site (i.e., it did not go to any subscribers). The laser being used to drive the fiber line was not specified to drive the entire length of 22 kilometers; it would normally be set up to drive a maximum of 12 kilometers and thus a reduction of Carrier-to-Noise ratio due to the longer fiber length incurred. Reasonably good BER numbers (4.7E–05) were obtained with the 256-QAM signal. The signal was set to be –5 dB lower in overall power compared to the adjacent NTSC signals and was bandlimited to be confined to the 6 MHz channel spacing by filtering the input to the upconverter with a standard Vestigial SAW filter centered at 43.7 MHz. **Figures 5** and **6** show the receive signal spectrum and constellation, respectively.

In Phase Two, a 256-QAM signal was transmitted through a "live" (i.e., on-the-air) cable system. The signal was transmitted through 23 trunk amplifiers, plus a high gain bridge amplifier (**Figure 7**). The BER of the 256-QAM signal was recorded at 4.3E–05 and 64-QAM was nearly error free at 8.2E–10. Both results were better than those obtained over the fiber link. The CNR of the channel was measured to be approximately 48 dB (~41 dB SNR). A spectral plot of the received digital signal, along with the adjacent aural and video carriers, can be seen in **Figure 8** with a photograph of the real-time 256-QAM constellation shown in **Figure 9**.

The Phase Three test consisted of demodulating the 256-QAM signal at a test subscriber site (within Manhattan Cable's building) followed by splitting to the various subscribers. The received BER was 1.2E–05.

The results of the tests are summarized in **Table 3** including the approximate link margins in parentheses (assuming an uncoded BER ~1.0E–03 is required with a coded system). Note that the tested channels were <u>not</u> SNR limited but rather phase-noise and filter distortion limited.

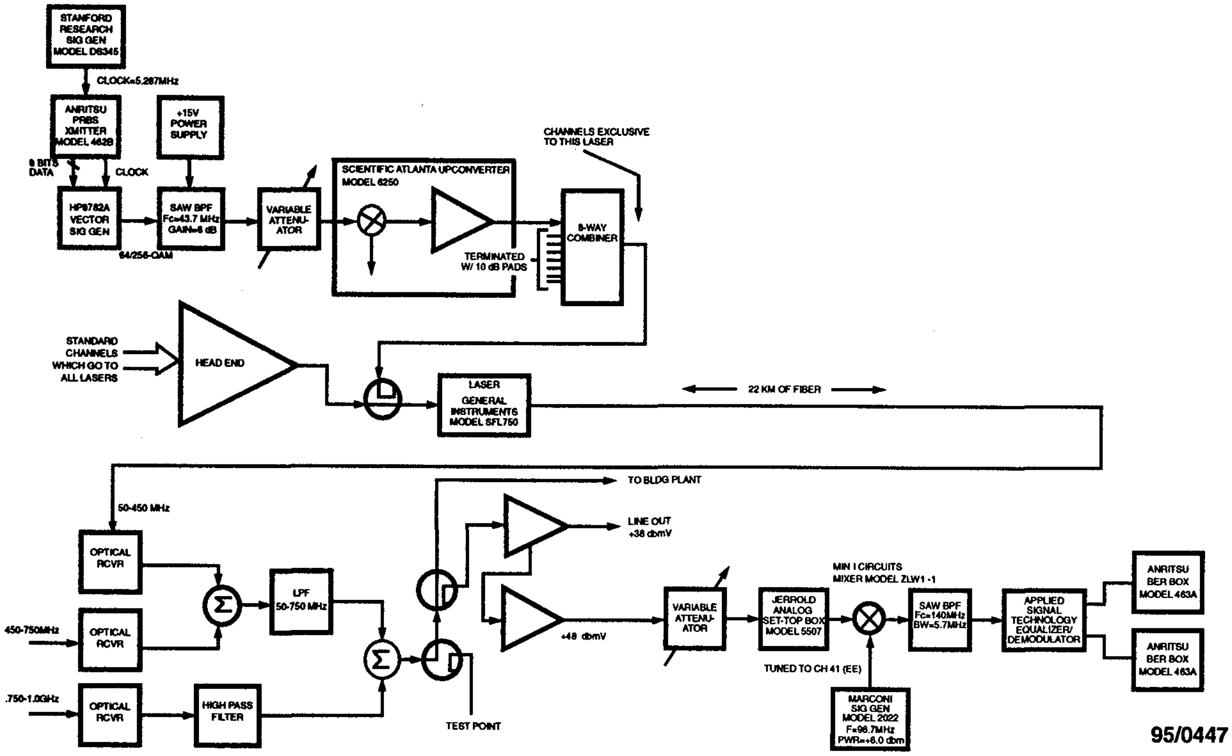



**Figure 4. Phase One Test: 22 km of Fiber**

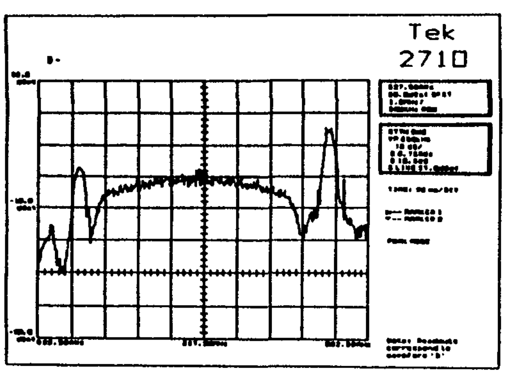



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**Figure 5. Phase One Test: Signal Spectrum**

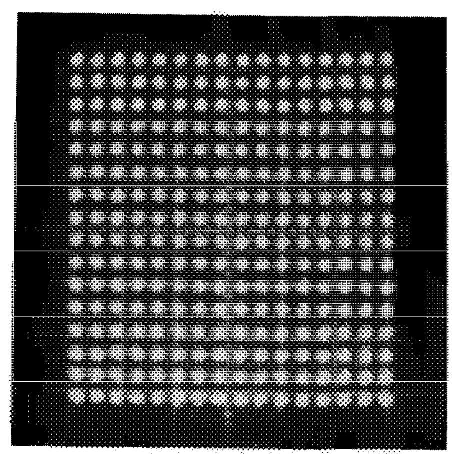

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**Figure 6. Phase One Test: Signal Constellation**

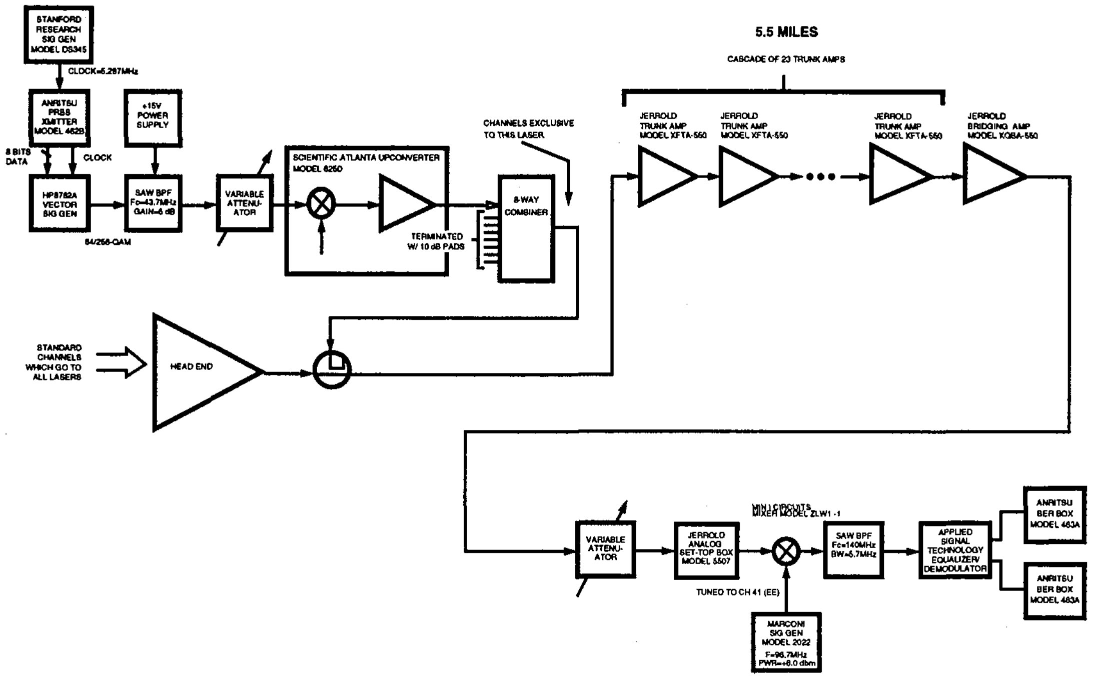


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**Figure 7. Phase Two Test: 23-Trunk Amp Channel**

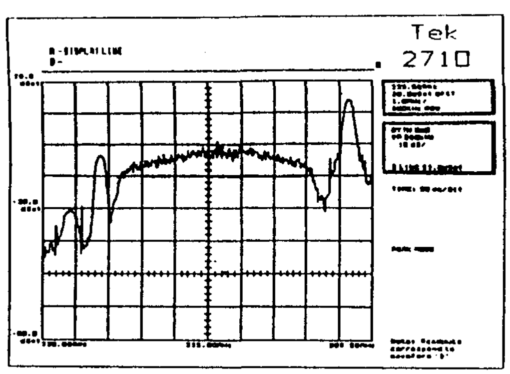


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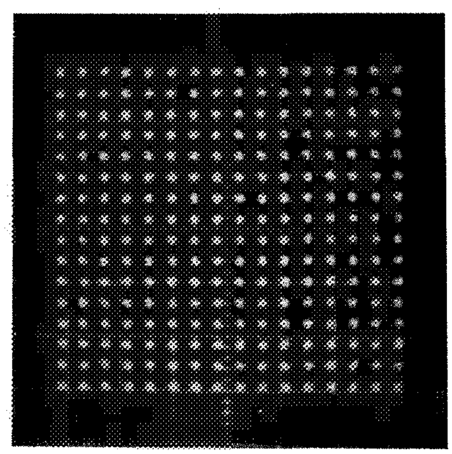
**Figure 8. Phase Two Test: Signal Spectrum**



95/0452P

**Figure 9. Phase Two Test: Signal Constellation**

**Table 3: Summary of Bit Error Rate Results (Uncoded)**

| Test Condition | 256-QAM BER | 64-QAM BER |
|---|---|---|
| Fiber Link | 4.70E–05 | 1.85E–08 |
| | (2 dB) | (5.5 dB) |
| Coax Run | 4.30E–05 | 1.85E–08 |
| | (2 dB) | (6.5 dB) |
| Subscriber | 1.20E–05 | Not Measured |
| | (3 dB) | |

The next phase of the test will include broadcasting four channels of NTSC quality compressed digital video over one 6 MHz cable channel. Transmission will utilize OmniBox's proprietary compression technology, DVC™ (Digital Vector Compression), with 256-QAM and Forward Error Correction.

## An Alternative Architectural Approach

As consumer price pressures assert their power on modem designers, the functional blocks which occupy the largest amount of real estate will come under increasingly close scrutiny. As mentioned previously, the equalizer functional block represents the top candidate for re-examination, occupying up to 40–50 percent of the demodulator die area. To this end, alternative equalizer structures were explored and a new architecture was chosen based upon signal processing considerations associated with the other functional blocks in the system and their relationship to video data rates and TV IF sampling methodologies. This architecture was arrived at as a result of an engineering trade-off between sufficient performance and the economic utilization of silicon. It is not the intent of this paper to examine the precise implementation aspects of the proposed design, but rather to state its existence, associated economy with respect to computational complexity and die area, and performance merits relative to the T/2-spaced prototype.

As a first order basis of comparison, consider a 32-tap, linear, T/2-spaced equalizer. It has been shown **[2,8]** that this is a reasonable length to span for most cable environments that might be encountered. In order to compute one output, there are 32 taps x 4 FIR structures (complex equalizer)=128 multiply accumulates (MACs) per output. Assuming a data rate of 5 Mbaud, this corresponds to 5Mx128=640M (MAC ops/sec). For an equalizer span which is equivalent to 32 T/2-spaced taps, the proposed design is capable of reducing the computational complexity by a factor of roughly one-third. In addition, die size comparisons with an existing T/2-spaced design have revealed an equalizer real estate savings of greater than 30 percent. Assuming that this architecture performed well in simulated cable environments with associated multiple impairment scenarios involving phase noise, microreflections, and Gaussian noise, a strong motivation to move toward a VLSI implementation would be warranted. Two computer models of the proposed architecture were created. The first model consisted of a floating point VAX simulation with reasonable first order quantization effects included. The second model consisted of a hardware specific fixed point COMDISCO simulation. In both models performance was characterized over single and multiple impairments for both 64- and 256-QAM. Throughout the design and simulation process, results were consistently being reviewed and compared to the T/2 prototype results in order to gauge the efficiency of the trade-off between performance and economy. The question continually posed to the design team was: "How did the T/2-spaced prototype perform in this scenario?"

## Simulation Results

A computer model of the advanced architecture was generated in floating point and simulated on a VAX with reasonable first order quantization on critical parameters such as the data and equalizer tap weights. The functions modeled include: digital downconversion from TV IF to baseband, baud timing recovery, carrier recovery, equalization, and symbol decisions. The block diagram of the simulated environment is shown in **Figure 10.** A 20 percent Square Root Raised Cosine (SRRC) transmit filter was assumed throughout. The impairments modeled included microreflections, additive Gaussian noise, phase noise, and carrier offsets. These impairments were first added individually, and the equalized SNR was measured. Finally, the impairments were applied to the channel as a group. The results for 64-QAM are shown in **Table 4**. Laboratory tests involving these same impairments were then conducted for purposes of comparison using

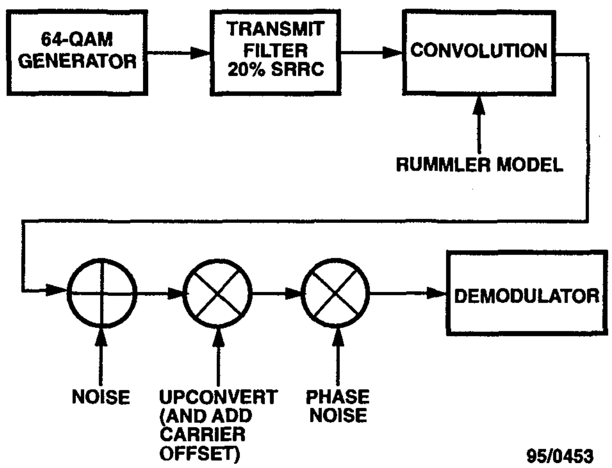



**Figure 10. System Simulation Block Diagram**

the Applied Signal Technology Model 242 (T/2-spaced) prototype. The results of these T/2 hardware tests are also shown in **Table 4**. Similar results for 256-QAM are shown in **Table 5**.

**Figure 11** shows the spectrum of the 64-QAM signal just before equalization for the multiple impairment scenario shown at the bottom of **Table 4. Figure 12** shows the constellation prior to demodulation. This scenario includes a microreflection with a relatively short delay of 500 nsec and a ray amplitude which is –10 dB down from the signal of interest. Note the deep notch in the center of the frequency response in **Figure 11** caused by the echo.This notch is due to the reflected ray summing 180 degrees out of phase with the signal of interest. The equalizer must build the inverse of this response in order to equalize the channel, as it does in **Figure 13**. A magnitude plot of the converged equalizer weights is included in **Figure 14**. The highest peak on the left represents the tap location where the impulse response of the equalizer was initialized. The second highest peak to the right signifies the location of the zero which the equalizer created to equalize out the effects of the pole caused by the echo. This second peak is –10 dB down from the first peak, which is precisely what one would expect since the echo is –10 dB down from the transmitted signal. Finally, **Figure 15** shows the demodulated 64-QAM constellation.

**Table 4: 64-QAM Performance Comparison of Simulated Architecture with Prototype T/2 Architecture**

| | Multipath | CNR (dB) | Phase Noise (dBc/Hz @ 10 kHz Offset) | Carrier Offset (kHz) | Equalized SNR (dB) |
|---|---|---|---|---|---|
| Simulation Model | None | No Noise | None | 0 | 38 |
| Model 242 (T/2) | None | No Noise | None | 0 | 37.1 |
| Simulation Model | –10 dB @ 500 nsec | No Noise | None | 0 | 36.8 |
| Model 242 (T/2) | –10 dB @ 500 nsec | No Noise | None | 0 | 36.8 |
| Simulation Model | None | 28 | None | 0 | 26.5 |
| Model 242 (T/2) | None | 28 | None | 0 | 26.8 |
| Simulation Model | None | No Noise | –72 | 0 | Not Tested |
| Model 242 (T/2) | None | No Noise | –72 | 0 | Not Tested |
| Simulation Model | –10 dB @ 500 nsec | 28 | None | 0 | 25.8 |
| Model 242 (T/2) | –10 dB @ 500 nsec | 28 | None | 0 | 26.7 |
| Simulation Model | –10 dB @ 500 nsec | 28 | –72 | 100 | 24.8 |
| Model 242 (T/2) | –10 dB @ 500 nsec | 28 | –72 | 100 | Not Tested |

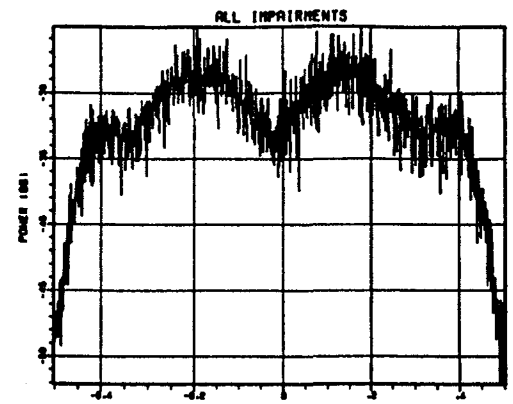



**Figure 11. 64-QAM Input Signal Spectrum: Multiple Impairment Scenario, Table 4**

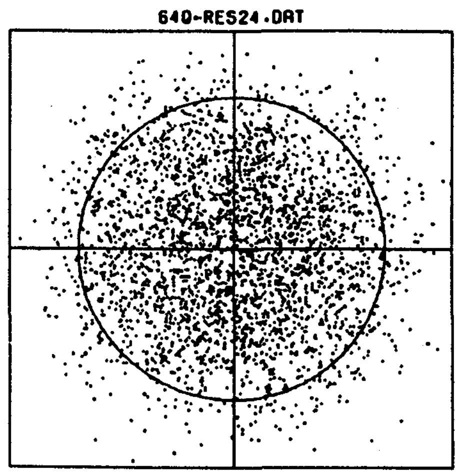



**Figure 12. 64-QAM Input Signal Constellation: Multiple Impairment Scenario, Table 4**

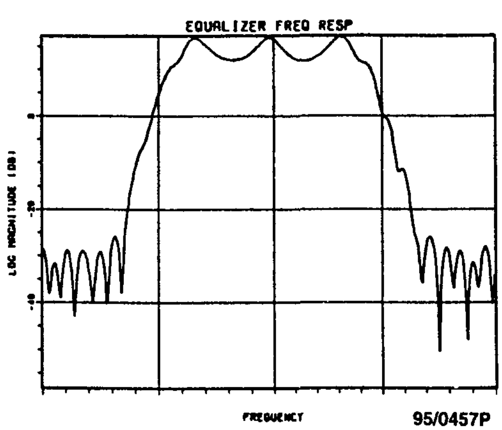



**Figure 13. Resultant Equalizer Frequency Response: Multiple Impairment Scenario, Table 4**

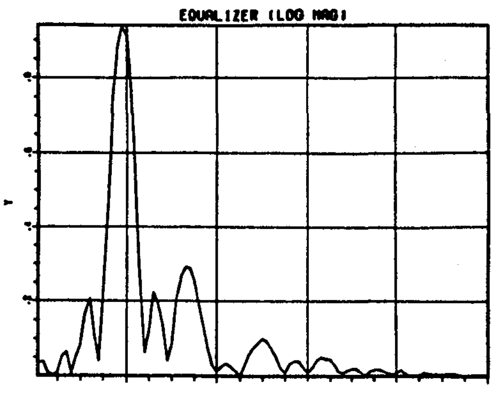



**Figure 14. Resultant Magnitude of Equalizer Weights: Multiple Impairment Scenario, Table 4**

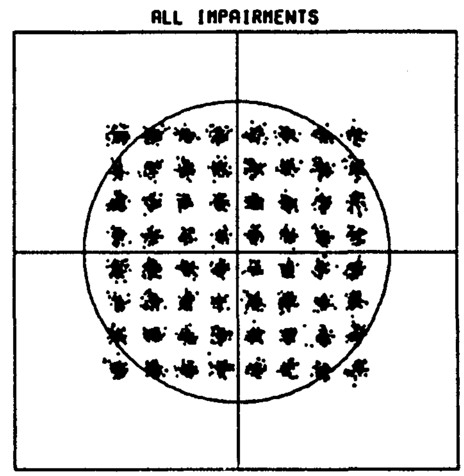



**Figure 15. Equalized and Demodulated 64-QAM Constellation: Multiple Impairment Scenario, Table 4**

Similar simulations were run with weaker reflected rays (–18 dB) and much longer delays on the order of 2 µsec. **Table 5** shows the results of a longer delay echo on 256-QAM. **Figures 16–18** correspond to this case. **Figure 16** shows the frequency response of the received signal prior to equalization. Note the "scalloping" that results in the amplitude of the signal spectrum. Larger notches such as those simulated in **Table 4** do occur, but are extremely improbable on real cable systems for a delay range of 1–2 µsec as has been shown in **[8]**. **Figure 17** shows the final equalized constellation. **Figure 18** shows the frequency response built by the equalizer after convergence. Comparison of this frequency response to that of **Figure 16** reveals that the equalizer response "enhanced" or amplified the notch energy in order to equalize the channel

**Table 5: 256-QAM Performance Comparison of Simulated Architecture with Prototype T/2 Architecture**

| | Multipath | CNR (dB) | Phase Noise (dBc/Hz @ 10 kHz Offset) | Carrier Offset (kHz) | Equalized SNR (dB) |
|---|---|---|---|---|---|
| Simulation Model | None | No Noise | None | 0 | 36 |
| Model 242 (T/2) | None | No Noise | None | 0 | 35.8 |
| Simulation Model | –18 dB @ 2 μsec | No Noise | None | 0 | 33 |
| Model 242 (T/2) | –18 dB @ 2 μsec | No Noise | None | 0 | 33.8 |
| Simulation Model | None | 33 | None | 0 | 31.5 |
| Model 242 (T/2) | None | 33 | None | 0 | 31.2 |
| Simulation Model | None | No Noise | –78 | 0 | 33 |
| Model 242 (T/2) | None | No Noise | –78 | 0 | Not Tested |
| Simulation Model | –18 dB @ 2 μsec | 33 | –78 | 100 | 28.7 |
| Model 242 (T/2) | –18 dB @ 2 μsec | 33 | –78 | 100 | Not Tested |

response. In both of the aforementioned cases the equalized constellations (**Figures 15** and **17**) exhibit the residual effects of Gaussian noise ("fuzziness" of the constellation points) and phase noise, i.e., "arcing" of constellation points in a direction perpendicular to the imaginary line between the point and the constellation origin. Note that the simulated results are within 1 dB of the T/2 prototype performance over all individual impairments and within 1 dB for the ensemble of impairments.

Finally, the advanced architecture was simulated against 64-QAM echo scenarios which were similar to those encountered in the CableLabs testing of October of 1994 (see **Table 3**). Note that these tests were run with the demodulator employing an equalizer with 64 taps initialized at tap 16. The goals of the simulations were twofold. First, we sought to determine the performance hit incurred by utilizing an equalizer which spanned the equivalent of 32 taps. Secondly, the desired performance degradation should include any effects associated with the advanced architecture. The results of these simulations are shown in **Table 6**. Three microreflection delays were simulated: 0.5, 1.0 and 2.0 microseconds. In each case the power of the echo relative to the desired signal was increased until acquisition was not longer achievable. Note that the advanced architecture matches the performance of the T/2-prototype to within 1 dB for echoes with time delays which are less than 1 microsecond. The performance hit becomes more noticeable at the longer delay of 2 microseconds. In this case the equalizer acquired an echo which was –11.5 dB from the desired signal. Note the resultant constellation in **Figure 19**. This represents a performance hit of approximately 4.5 dB relative to the 64-tap T/2-spaced hardware prototype. However, the results are still quite promising when one compares this 2 microsecond, –11.5 dB ray with the statistical results of measurements taken over a wide variety of cable systems **[8]**. To reiterate, 99 percent of all subscriber sites measured delay times less than 1.28 microseconds and echo amplitudes less than –19 dB below the desired signal.

## SUMMARY

The results of extensive testing performed on a 64/256-QAM demodulator prototype employing a T/2-spaced equalizer were presented. Testing covered a multitude of cable impairments in both a laboratory and "real world" environments. The majority of the laboratory testing was done at the Advanced Television Test Center (ATTC). Testing involving a "live" cable system was performed in the New York City field trial over Time Warner's

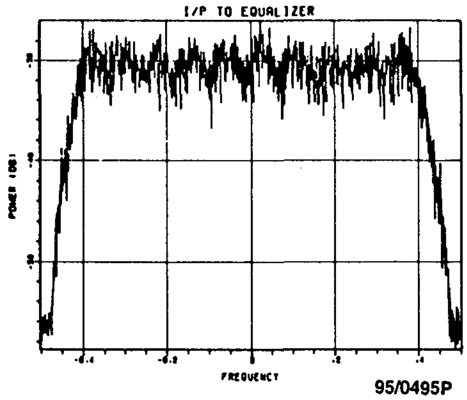



**Figure 16. 256-QAM Input Signal Spectrum: Multiple Impairment Scenario, Table 5**

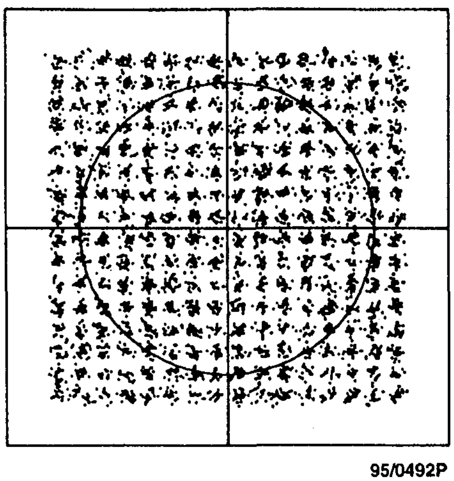



**Figure 17. Equalized and Demodulated 256-QAM Constellation: Multiple Impairment Scenario, Table 5**

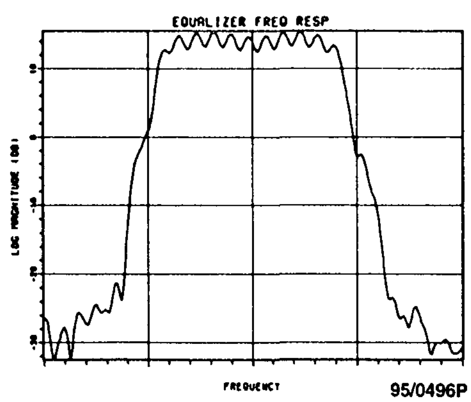



**Figure 18. Resultant Equalizer Frequency Response: Multiple Impairment Scenario, Table 5**

**Table 6: 64-QAM Echo Simulations vs. Prototype Performance**

| | **Simulation [1]** | **T/2 Prototype [2] Tested by CableLabs** |
|---|---|---|
| **Delay (μs)** | **Acquisition (dB relative to signal)** | **Acquisition (dB relative to signal)** |
| 0.5 | –5 | –5 |
| 1 | –7 | –6 |
| 1.5 | | –6 |
| 2 | –11.5 | –7 |
| 2.5 | | –10 |
| 3 | | –11 |
| 3.5 | | –11 |
| 4 | | –11 |
| 4.5 | | –10.5 |
| 5 | | –20 |

(1) Time span of equalizer equivalent to 32 T/2-spaced taps, initialized at approximately tap 8

(2) 64 taps, initialized at tap 16

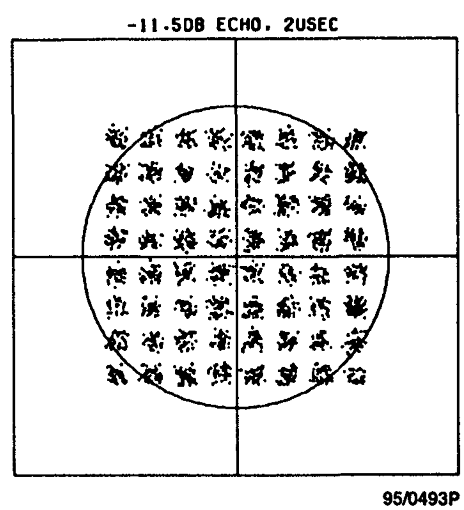



**Figure 19. Equalized and Demodulated 64-QAM Constellation: –11.5 dB Echo at a Delay of 2 μsec**

system. All performance testing indicates that QAM is an operationally effective means of high speed data transmission for digital CATV and that a QAM receiver employing blind equalization and carrier recovery as embodied in the T/2-spaced prototype architecture is sufficiently robust for a wide range of realistic impairment scenarios.

An alternative to the T/2-spaced architecture was suggested as a means of decreasing the equalizer computational complexity, and therefore die real estate, by a factor of roughly one-third. A computer model of the advanced architecture was created and performance was simulated against both single and multiple impairment scenarios.The results of the simulations for the advanced architecture were very close (typically within 1 dB) to that of the T/2-spaced prototype for equivalent equalizer time spans. The results were very encouraging based upon the success of the T/2 prototype at the Advanced Television Test Center and in live cable systems. The positive impact of these simulations led to a second phase of simulation modeling for the proposed architecture. In this second phase, a model was created in COMDISCO which incorporated a hardware-specific approach and included detail down to the flip-flop level.

## ACKNOWLEDGMENTS

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# Traffic Management for Highly Interactive Transactional System

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Time Warner Cable

Abstract

*Management of heterogeneous real-time traffic is a key subject in the development of a full service network for the delivery of multi-media assets. The requirements of MPEG compressed material (currently constant bit rate) are contrasted with the requirements for other multi-media assets such as graphics, data and programs (variable bit rate). Traffic management in the Orlando Full Service Network™ is described in some detail, and is used to demonstrate how both types of traffic (i.e., compressed video and audio streams multiplexed together with IP data streams) can be accommodated on a single ATM network. The design and management of such a network is discussed in some detail, using the lessons learned from the Orlando trial to illustrate the allocation of traffic across a switched network, from the server complex, over the hybrid fiber/coax broadband access, to the homes.*

## Introduction

The Time Warner Full Service Network™ was designed to provide broadband digital interactive services over the hybrid fiber/coax network. It represents a pioneering step in the development of technology and marketing to introduce a full range of new digital services to home consumers, ranging from interactive TV to telephony, data communications and utility metering. The Orlando Full Service Network trial represents the first instance of the concept, and it is designed to support 4000 subscribers, with an expected simultaneous load estimated at 1000 full-motion audio-video streams. The Orlando Full Service Network is based on a switched, distributed architecture, where individual audio and video media streams - coming from a server complex at the head-end - are switched to the respective Home Computing Terminal (HCT) at the customer's residence. The software to control the operation of the media servers and the implementation of the applications forms a complex distributed computing environment that requires its own extensive communications capabilities to exchange application code and other data.

The Orlando Full Service Network uses a single Asynchronous Transfer Mode (ATM) node to support transmission of application code and data, and compressed MPEG audio and video streams. These traffic types have different characteristics and quality of service requirements [1]:

- Application code and data do not have stringent delay requirements, hence retransmission is possible to achieve a reliable network layer, and some packet losses at the physical/link levels are acceptable.
- MPEG audio and video streams are real-time media, hence they have stringent delay requirement and retransmission is prohibited. Also, they do not tolerate packet losses well. Therefore, MPEG audio and video streams must be accurately rate-controlled to prevent buffer overflow in the network and at the MPEG player, and forward correction methods are necessary to improve reliability of network transport.

It is worth noting some of the assumptions and requirements that are specific to the design of the Orlando Full Service Network trial.

- The network is required to support 1000 simultaneous MPEG audio-video streams at approximately 4 Mbit/s per stream. Hence 4 Gbit/s of forward capacity is required for this purpose. Additional capacity is required for application code and data download.

- The network must provide sufficient bandwidth to allow signaling with low-latency in the reverse direction (from the subscriber to the network). However much less reverse bandwidth is required than forward bandwidth and it was recognized that the network would be highly asymmetric in nature. In practice, approximately 95% of the bandwidth is in the forward direction and 5% in the reverse direction.

- The system would require more than one media server for reasons of capacity and fault tolerance.

- The access hybrid fiber/coax network is composed of multiple access sub-networks (called neighborhoods), based on the allocation of approximately 300-500 homes on a common coaxial plant for each fiber node.

The above assumptions and requirements led to an ATM paradigm. To allow for maximum flexibility, the servers need to be connected to the neighborhoods by means of a switching network that can establish connections from any server to any customer. An ATM switching approach was the only viable approach given the multi-gigabit bandwidth requirements.

ATM is a transport, multiplexing and switching technology based on fixed-length cells of 53 bytes that carry both the payload (i.e., the data to be delivered) and the header. The 5-byte header contains addressing information for each cell consisting of a Virtual Path Identifier and Virtual Circuit Identifier (VPI/VCI) pair for each network link. A given connection is established by setting the proper VPI/VCI pair associated with the source and destination. Hence, a given stream of ATM cells can interleave data from many different connections by carrying cells with the appropriate VPI/VCI pairs.

## Network Design

Figure 1 is an overview of the Orlando FSN network. The network design was driven by the following key facts:

- ATM signaling standards and implementations for Switched Virtual Connection (SVC) were immature, and a switching network based on Permanent Virtual Connections (PVCs) was chosen.

- The Constant Bit Rate (CBR) MPEG video streams were mapped into Constant Bit Rate CBR ATM connections. Reserved bandwidth was guaranteed for each video/audio connection.

- Internet Protocol (IP) was selected as the data communications network protocol to allow the re-use of existing networking software for the data communications between network endpoints.

The logical routing and addressing of the various information streams are shown schematically in Figure 2. In the forward direction, the information from the server complex consists of multiple OC3 streams, each containing multiple ATM connections. The ATM switch routes each ATM connection to its appropriate output port operating at DS3 rate. The various DS3 streams are modulated at their corresponding carrier frequencies (see later section for RF spectrum assignment discussion), and delivered over the hybrid fiber/coax to the respective HCTs. The HCT at each individual home receives multiple streams of ATM cells at DS3 rates. In order to select the correct information that has been switched to its specific application, the HCT first tunes its demodulator to the correct RF frequency, and then it selects the individual ATM stream based on its VPI/VCI pair.

In the reverse direction, multiple ATM connections at DS1 rate from the HCTs are merged in each Neighborhood Area Node (NAN) using a TDMA scheme to avoid contention. These ATM connections are multiplexed up to DS3 rates, and then routed by the ATM switch to the appropriate server interface at the headend. A Connection Management Module, part of the distributed control system, is responsible for managing the establishment of the connections between the server complex and all the users HCTs.

As described, the network is asymmetric and the following overview will cover a description of the forward and reverse directions. This contrast is worth noting as most communications protocols assume a bi-directional (i.e., symmetric) physical facility. Considerable work was necessary to adapt these protocols to uni-directional facilities used in the Full Service Network.

### Forward Direction

Eight media servers (SGI Challenge XLs) are connected to disk vaults using fast and wide SCSI-2 interfaces. The vaults can be configured to provide a total of 1.7 terabytes of media storage capacity or about 500 movies.

The media servers are connected to a GCNS-2000 ATM switch supplied by AT&T. A total of 48 SONET OC3 links provide 5184 Mbit/s of forward payload bandwidth. Each OC3 provides a net payload of approximately 108 Mbit/s (see later section on OC3 for more explanation).

The media servers are also interconnected with a FDDI ring. This ring is used to transfer media content to the disk vaults and to collect billing records from the servers. A separate FDDI ring was used only to expedite development, and the ATM switched network will be used to support all communications needs in future full service networks.

The ATM switch is connected to a bank of QAM-64 modulators supplied by Scientific Atlanta. There are 152 uni-directional DS3 links to provide a total of 5600 Mbit/s of payload capacity from the ATM switch to the neighborhoods. Each DS3 provides 36.86 Mbit/s of payload capacity after ATM and PLCP overheads are taken into account.

The QAM modulator outputs are defined at carrier frequencies from 500-735 MHz spaced at 12 MHz. This allows the outputs to each neighborhood to be combined into a composite RF signal. Conventional analog television channels in the range 50-500 MHz (spaced at 6 MHz) are also provided. The RF spectrum assignment diagram is shown in Figure 3.

The composite RF signal from 50-735 MHz is then used to amplitude modulate a laser. The laser is coupled to a single-mode fiber which takes the signal out to the neighborhood about 10 miles away. At the neighborhood the optical signal is converted back into the RF domain by the Neighborhood Area Node (NAN) and used to feed a coaxial feeder network which passes about 500 subscribers.

The RF signal enters the subscriber residence and feeds the Home Communications Terminal (HCT) or set-top. The HCT is a powerful RISC-based multi-media computing engine with video and audio decompression and extensive graphics capabilities.

### Reverse Direction

The HCT transmits a QPSK-modulated signal in the 900-1000 MHz band. Note that this high-split RF for the reverse direction is a novel feature of the Orlando FSN. Reverse carrier frequencies are defined at a spacing of 2.3 MHz. The QPSK channel operates at DS1 rates, providing a net data rate of 1.152 Mbit/s after accounting for ATM overhead. Each reverse channel is slotted using a Time Division Multiple Access (TDMA) scheme. This allows a single reverse channel to be shared among a number of HCTs. The slot assignments are made at the head-end and sent to the HCT over a forward channel such that no more than one HCT is enabled to transmit in any given slot. By default, each HCT has access to a constant bit rate ATM connection with a bandwidth of 46 Kbit/s. More importantly, the access latency of a typical packet is 25 ms worst-case.

The reverse channels from a neighborhood are transported by the coax plant back to the Neighborhood Area Node (NAN). At this point, the reverse spectrum is used to modulate a laser, which is coupled to a single mode fiber. Separate fibers (in the same sheath) are used for the forward and reverse directions.

At the head-end, the optical signal is first converted back into the RF domain and the fed to a bank of QPSK demodulators. These convert to cell-stream into a ATM-format DS3. The mapping is standard but the DS3 is a uni-directional link.

The outputs of the demodulators are combined by seven ATM multiplexers (supplied by Hitachi). A standard, bi-directional ATM-format DS3 is used to connect each multiplexer to the ATM switch.

### ATM Addressing

The ATM switch and multiplexers are configured with a mesh of Permanent Virtual Connections (PVCs).

In the forward direction, Virtual Paths are configured from each OC3 port to each DS3 port. This allows the server to address any Forward Application Channel (FAC) by selecting the appropriate VPI. The HCTs tuned to a FAC ignore the VPI and reassemble connections based on VCI. Thus we have a two-level switching hierarchy, VP switching in the ATM switch and VC switching in the neighborhood.

In the reverse direction, the multiplexers perform a traffic aggregation function from DS3 to DS3 rates. A mesh of virtual paths is provisioned from the multiplexer DS3 ports to the server OC3 ports. This allows any HCT to send cells to any server port by using the appropriate VPI.

### Connection Management

The allocation of VP and VC identifiers and of network bandwidth is performed by the Connection Manager. The Connection Manager is a distributed set of processes than run on the media servers. In response to an application request for a connection with a given quality of service, the Connection Manager determines a route, allocates connection identifiers and reserves link bandwidth. The connection identifiers are returned to the server and client applications at the media server and HCT respectively.

### ATM Mapping

The mapping of higher-level protocols into ATM virtual channels is outlined in Figure 4. Classical IP mapping over ATM [2] is used with some modifications to support uni-directional virtual channels. MPEG system layer streams are mapped directly into AAL-5, with a separate virtual channels for audio and video. The emission rate of each virtual channel is carefully rate-controlled in the server interface to minimize queue lengths in network buffers [3]. At present, no isochronous streams are supported.

## Bandwidth Allocation

From its inception, the Orlando FSN was designed to serve 4000 subscribers with a maximum load of 25% of those subscribers simultaneously accessing interactive, full-motion services, generating a capacity requirement of 1000 video streams. Investigation of available MPEG compression technologies convinced us that a 3.5 Mbit/s compressed video data rate was required for high-quality pictures from a movie source (as good or better than S-VHS). The audio compression rate was chosen to be 384 Kbit/s to provide high quality stereo and matrixed surround sound. This yields to a total of 3.949 Mbit/s per stream after allowing for system layer overhead. (This is rounded to 4 Mbit/s per stream for the calculations in this paper). In addition, any single neighborhood was designed to support a peak load generated by 40% of subscribers simultaneously accessing interactive, full-motion services.

The entire OC3 bandwidth cannot be used for movies on demand because some bandwidth must be reserved for messaging and application loading. (Recall that each audio-video stream requires approximately 4 Mbit/s for audio and video).

In the current design, 4160 Mbit/s, of the 5184 Mbit/s of total OC3 capacity, is allocated for up to 1040 Audio-Video (AV) streams. The remaining 1024 Mbit/s is reserved for messaging and application download. This is almost 20% of total OC3 bandwidth. Why is so much bandwidth reserved for this purpose?

### DS3 Bandwidth Allocation

The answer is apparent if we work backwards from the DS3 Forward Application Channels (FAC). Each FAC contains a number of sub-channels which are actually ATM Virtual Channels (VCs). These Virtual Channels are created dynamically by the Connection Manager and can carry audio, video or data traffic. This is illustrated in Figure 5. (Note that for simplicity the audio and video channels are grouped together (AV1 or AV2) in the diagram). The diagram illustrates 6 FACs but in reality a neighborhood will typically carry 9 or more FACs (depending on subscriber count).

Each FAC carries two IP channels, Fast IP and Slow IP. (These are grouped and labeled simply 'IP' in the diagram). A FAC can also carry a number of Audio/Video channel-pairs (labeled AV1 or AV2). There are two video rates:

- AV1 requires approximately 4 Mbit/s to deliver 24 fps movie material and 384 Kbit/s audio.
- AV2 requires approximately 6 Mbit/s to deliver 30 fps video material and 384 Kbit/s audio.

### IP Channels

Two IP sub-channels exist on each Forward Application Channel; a 'slow' IP channel of 0.714 Mbit/s and a 'fast' IP channel of 8 Mbit/s. Two IP channels are used so that the HCT can mask traffic on the Fast IP channel during normal operation thus saving CPU bandwidth that would otherwise be used for unnecessary receive processing.

Slow IP is used for general messaging between the servers and HCT. A 0.714 Mbit/s channel has proved to be more than adequate for this purpose.

Fast IP is used to download application code to the HCT when required; for example, when a subscriber enters the movie-on-demand venue. Timely application download is ensured by allocating sufficient bandwidth for the Fast IP channel (8 Mbit/s or 23.6% of the total FAC bandwidth). If a typical application contains 8 Mbits (or 1 MByte) of data, it will take approximately 1 second to download.

Two or more HCTs, tuned to the same FAC, may request application download at the same time. In this case, the Fast IP channel is shared between them and the download time is proportionately longer.

Each FAC has its own IP channels. Early in the design it was thought that a single FAC channel could be shared by all HCTs for download. Unfortunately, this is not possible because it would add significantly to the HCT response time due to the time needed to re-tune. For example, if an HCT were tuned to FAC3 and playing a movie, when the subscriber makes a request for home shopping, the HCT would have to re-tune to the download FAC to request the home shopping application, and then re-tune to FAC 3 to receive the home shopping AV stream. The re-tune time is approximately 200 ms, thus 400 ms of latency would be added. Worse still, during re-tune and application download the HCT can display **only** locally generated graphics and audio. This would place an unacceptable burden on the application developer who has to keep the user interested for typically 1.4 seconds with locally generated cover.

For these reasons, it was decided to allocate bandwidth in every FAC for Fast IP as well as slow IP. At 8 Mbit/s the Fast IP Virtual Channel consumes as much bandwidth as 2 AV1 (movie delivery) streams. When this decision was made, network capacity was increased to carry 1000 AV1 streams after taking IP overhead into account. Each FAC can carry 7 AV1 streams and so DS3 links were added to provide a total of 152 FACs. These provide capacity to carry 1064 (152 x 7) AV1 streams.

### OC3 Bandwidth

As previously stated, sufficient OC3 bandwidth must be reserved to support 1000 AV streams, each at 4 Mbit/s. This requires 4000 Mbit/s of bandwidth.

Each OC3 can carry 108 Mbit/s and there are 48 OC3 links, giving a total of 5184 Mbit/s. Thus the bandwidth available for IP is **1184** Mbit/s (5184 - 4000 ).

However, the total bandwidth required to support all 152 FACs for IP is **1325** Mbit/s given 8.714 Mbit/s per forward application channel. (The reason for this discrepancy is that there is more FAC bandwidth than OC3 bandwidth to allow for per-neighborhood peak loads of 40%).

To solve this discrepancy, we could add more OC3 bandwidth, increasing cost. Instead, an alternative approach was chosen; to use statistical multiplexing at the OC3 links and over-subscribe the available OC3 bandwidth. To do this effectively, the IP traffic must be **flow** controlled to prevent packet loss, and the IP channels must be **rate** controlled to prevent the IP traffic from exceeding the allowed connection bandwidth.

## IP Traffic Characteristics

The traffic on the Fast and Slow IP channels is bursty in nature. In particular, the Fast IP channel is only used while a HCT is loading an application. If we take a community of 4000 HCTs, the probability of a significant fraction of them simultaneously loading an application is low.

### Rate Control

Each OC3 card has a number of rate-queues which are implemented in the OC3 interface hardware. A rate queue allows ATM cells to be emitted at a constant rate. There are 5 rate queues defined in each OC3 card as follows (in rate order):

- 8 Mbit/s for Fast IP
- 5.56 Mbit/s for 30 fps MPEG compressed material (video)
- 3.56 Mbit/s for 24 fps MPEG compressed material (movies)
- 0.714 Mbit/s for Slow IP
- 0.393 Mbit/s for MPEG audio (MusiCAM)

The rate queues also have high or low priority. All high priority rate queues are serviced before the low priority rate queues. Within a priority class, rate queues are serviced in a round-robin scheme to provide virtual channels which have a constant cell rate.

The IP traffic is tagged as a low priority and the AV traffic as high priority. The AV traffic must be accurately rate-controlled (within about 1%) to prevent buffer overflow or under-flow in the HCT. However, the IP traffic need only be rate-limited; as long as the IP rate does not exceed the bandwidth allocated in the FAC there is no impact to the service except for greater latency in application download.

### Flow Control

The MPEG compressed streams are constant bit-rate, which means that their bandwidth allocation is always fully utilized by a constant stream of data. In contrast, the Fast IP and Slow IP channels are variable bit-rate. In fact a particular Fast IP channel may carry no traffic at all for minutes or even hours.

Statistical Multiplexing takes advantage of the averaging effect of many bursty traffic sources when sharing a constant bit-rate link. However, a flow-control mechanism is required to prevent packet loss. This can be seen if the following scenario is considered:

Assume 20 bursty traffic sources each with a maximum bandwidth of 10 Mbit/s are sharing a physical link with a maximum capacity of 100 Mbit/s. On average, less than 10 sources are active at the same time and there is no packet loss. However, when more than 10 sources become active, some packets must be discarded or buffered. Buffering is only a very short term solution at these bit-rates. 1 Mbit of buffering will only survive for 10 ms if all 20 traffic sources become active. If the traffic sources re-transmit discarded packets this effectively increases their offered load to the link. The link can become flooded with re-transmissions and little or no real traffic (i.e. packets carrying useful payload) will successfully traverse the link.

If we plot the above scenario as a graph, of effective throughput versus offered load, it would appear somewhat like the curve in Figure 6 labeled 'packet loss' [4]. As the offered load increases, throughput increases linearly until the link becomes congested. At this point if the load is increased further, the effective throughput falls dramatically as most of the link bandwidth is occupied by re-transmissions.

If sufficient buffering is available in conjunction with a flow control mechanism, the curve in Figure 6 labeled 'no packet loss' would be observed. In this case, the source waits for acknowledgment before sending more traffic and thus regulates its output. The FSN design allocates sufficient buffering at the server to ensure the 'no packet loss' case.

The effect of flow control is to reduce the rate of each source according to the total load on the shared link. There are a number of algorithms that have been developed which attempt to keep the offered load at, or below, the 'knee' shown in the graph. In particular, the TCP 'slow-start' algorithm adjusts the TCP window size according to the success or failure to transfer packets over the link. On startup, the window size is set to an initial value and is increased as acknowledgments are received indicating that the packets were successfully received at the far end. If a negative acknowledgment is received or a time-out occurs waiting for acknowledgment, the window size is reduced.

Using the same numbers as before, 1325 Mbit/s of IP source bandwidth is allocated to 1184 Mbit/s of link capacity, over-subscribing it by 12%. This is very conservative, and as experience is gained with the IP traffic profile, it is expected that much higher statistical multiplexing gain can be used with no perceptible impact to the system response time.

The TCP slow-start mechanism has been employed in the FSN design. Test results show the same download time as TCP implementations without the slow-start algorithm.

It is essential for this scheme to operate successfully that IP traffic cannot increase to beyond its rate allocation and contend with AV traffic. To help ensure this the IP traffic uses a low-priority rate queue. Test results show that the TCP traffic allocated to the low-priority rate queue does not interfere with AV traffic in the high-priority rate-queue.

## Conclusions

The Full Service Network uses a single ATM network to support transmission of application code and data and compressed MPEG audio and video streams. These have very different traffic profiles and quality of service requirements:

- Application code and data must delivered in a timely fashion, but some packet loss can be tolerated if a reliable transport layer is employed. Delay requirement is not stringent.

- MPEG audio and video streams do not tolerate packet loss well and must be precisely rate-controlled to prevent buffer overflow in the network and at the MPEG player. Stringent delay requirement and retransmission is prohibited.

A combination of hardware-based rate control and software-based flow control was successfully employed to provide statistical sharing of bandwidth to a large number of subscribers. Rate-control was also used to prevent output buffer overflow in the ATM switch when many AV streams are combined.

In future, larger networks with more general topology the techniques described may not be adequate to regulate the flow of IP traffic, and techniques such as fast buffer reservation [6] may be required. In addition, techniques to bound delay and jitter of multimedia traffic [7] may be required for larger networks.

## Acknowledgments

The work described in this paper was made possible with the help of many colleagues in Time Warner, AT&T, C-Cube, Scientific Atlanta and Silicon Graphics. In particular, Louis Williamson and Tom Speeter deserve special credit for their contribution.

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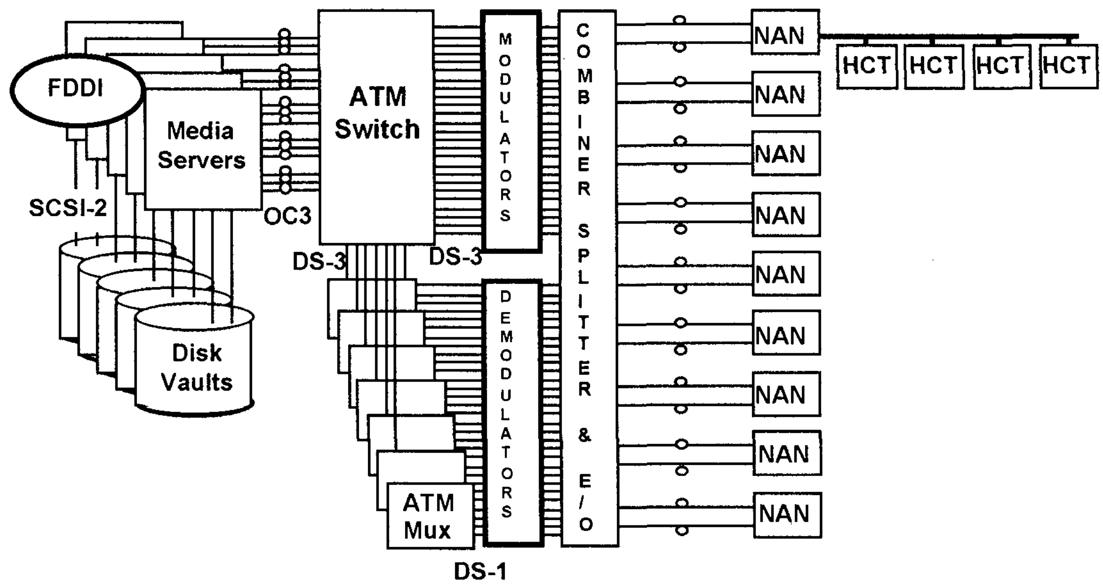


Figure 1. Full Service Network Overview

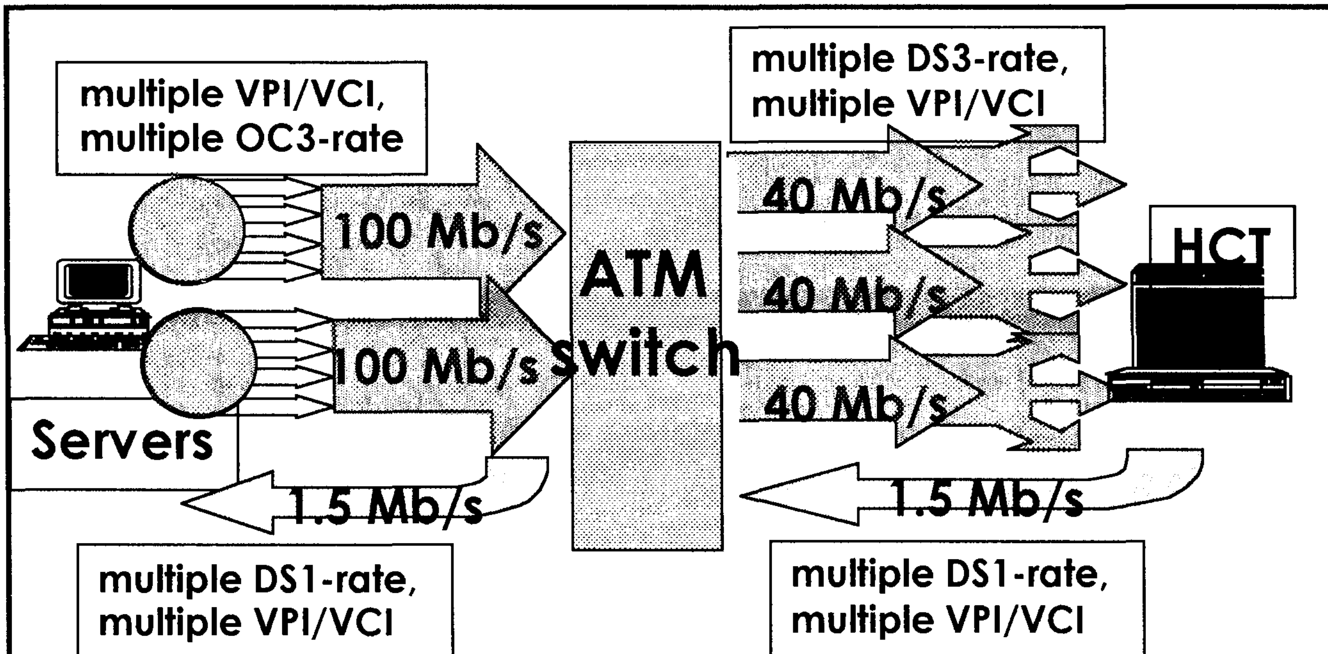


Figure 2: Logical ATM Connectivity

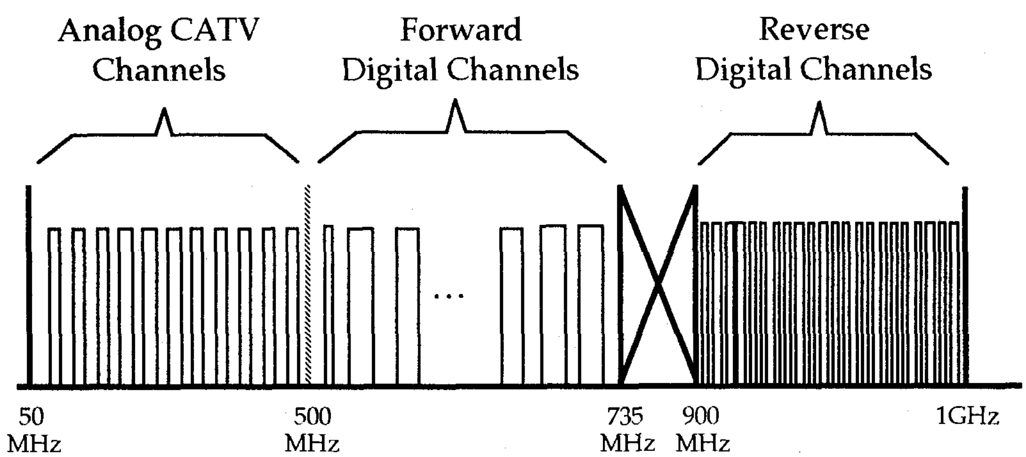


**Figure 3. Allocation of Analog and Digital Spectrum**

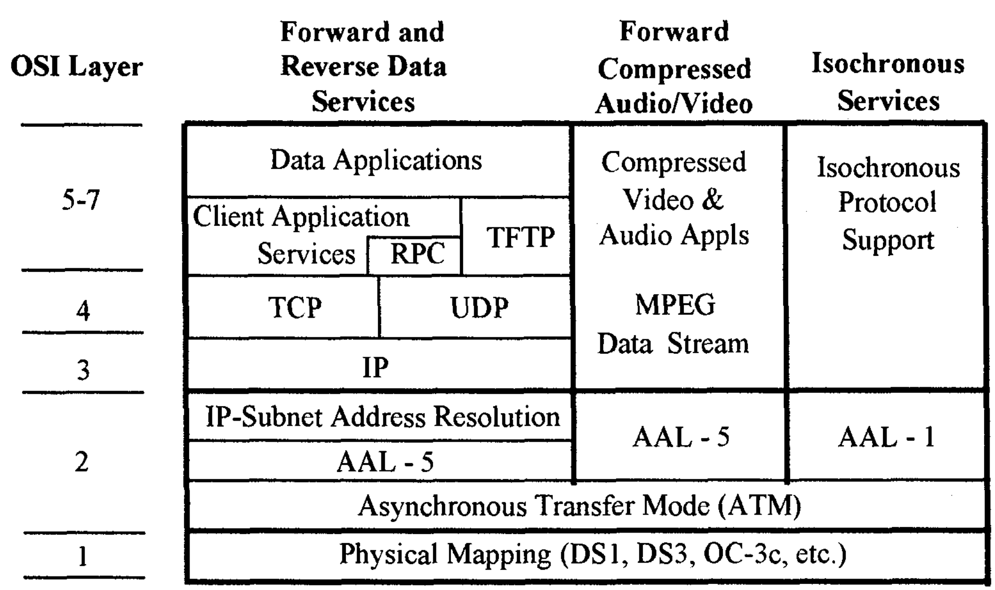


**Figure 4. Mapping of Higher-Level protocols into ATM**

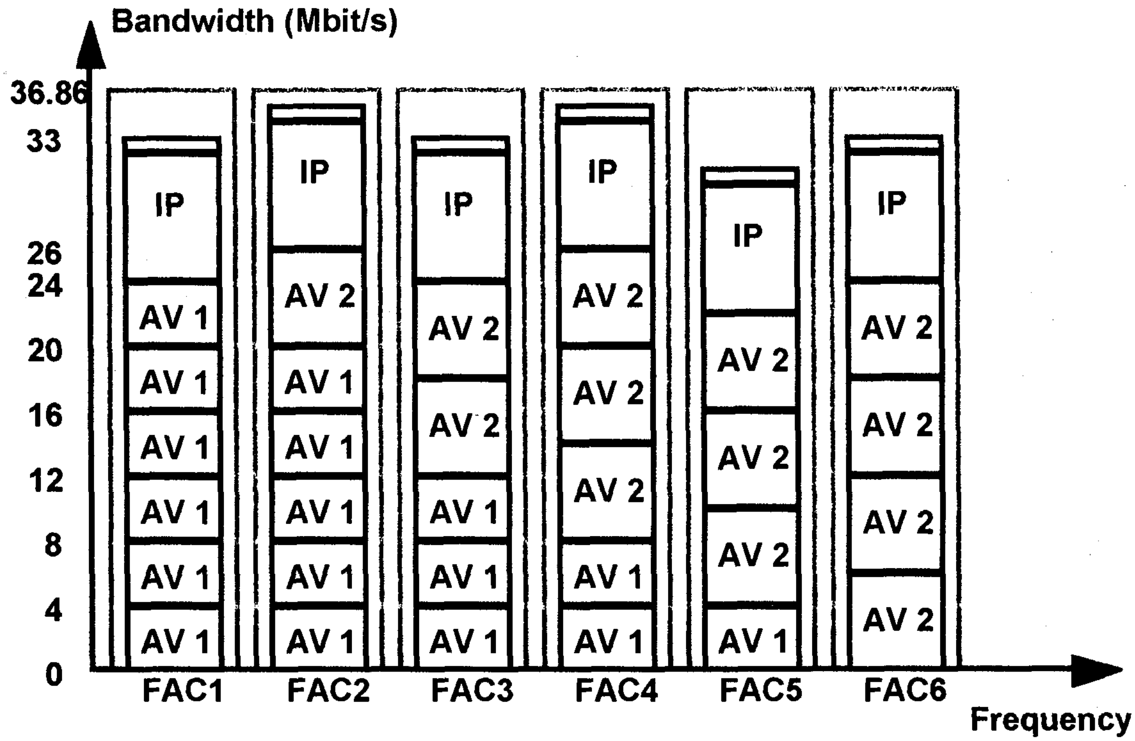


Figure 5. Bandwidth Allocation within Forward Application Channels

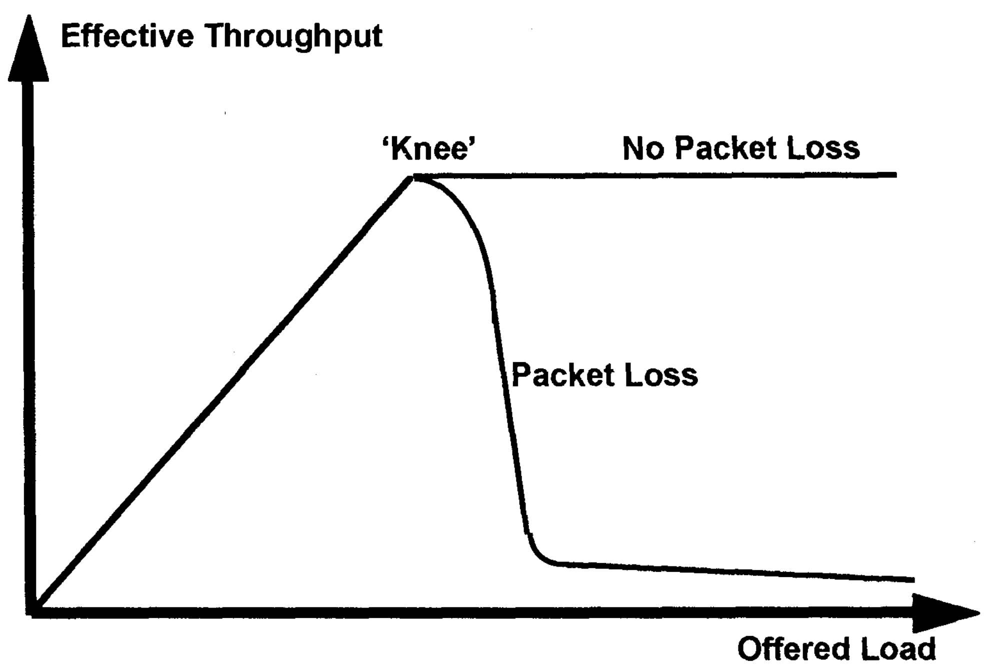


Figure 6. Effect of multiplexing many sources on a shared link

**Transmitting Power in the 90s:**
**Architectures and Systems**

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*Abstract*

*Powering is becoming an increasingly costly part of delivering communication services. With the advent of so much new technology, there are as many powering problems to accompany it. This paper deals with some of the problems associated with high power delivery and techniques to solve them. The bottom line is lower capital expenditure and lower operating costs.*

Anyone who has attempted to design a broadband cable system capable of delivering dial tone service has already discovered the enormous appetite for power that such a design requires. Short of deploying a power supply at every other corner with a minimum of six batteries per cabinet, a new solution must be found. Many of the designers are searching for new techniques and answers for this problem.

## The Problems

As cable systems have evolved, the powering needs have changed in step with many different elements of the broadband system. Power hungry amplifiers have changed the powering design such that for the first time, the network current ( the power in one single power feed direction ) has become the limiting factor. This coupled with the highly capacitive nature of the current consumption created by switching power supplies has created numerous problems for operating systems and power supply vendors. The actual current consumption provides an opportunity for the most improvement. Any line gear power supply will draw most of its peak current during the voltage peak of the waveform. In its worst case, a hypothetical 4 amp RMS load could produce peak currents of 24 amps. While this is alarming, the use of linear transformers in the older line gear resulted in a high loss of efficiency which was expended as heat in the amplifier housing, another situation that is not desirable.

Enter dial tone services. In addition to the power consumption of the line actives, a system that is delivering Dial Tone Service must also deliver 4 to 7 watts of power for each active dial tone customer to power a home device. The normal output wattage of a CATV UPS power supply is 900 watts. If we assume a 40% penetration of dial tone customers, the additional power required is upwards of 800 watts in a 500 home node. It is easy to imagine that with any actives, a single standard CATV product cannot deliver the powering needs of one fiber node.

When we consider the implications of 911 services on the hybrid system, it is important that the system be reliable. This reliability takes two different formats. The components must be redundant such that a component failure cannot stop the delivery of power. A failure must be serviceable so that it can be "hot swapped" while not producing an outage. The system must also provide a minimum of 8 hours of backup capability. Finally, there is a technology called "PCS" that will require 50 watts per transponder location. The net addition is somewhere between 150 watts and higher per node site. These challenges have led to what will be a drastic change in how we design and deliver power in the 90's.

## 90 Volt Powering

Anyone who has done design work knows that the biggest limiting factor in plant powering is the $I^2R$ losses of conductors, more notably coaxial cable and it's loop resistance. The industry fought this battle years ago and moved from 30 VAC powering to 60 VAC powering to increase the distance that power could be delivered over. Now with today's loads, it is time to increase that voltage again. The National Electrical Safety Code (N.E.S.C.) will allow up to 90 VAC on the systems for communications. However, the National Electric Code (N.E.C.) will only allow 60 VAC on the communication carrier to the home. They will however allow a secondary carrier to transmit up to 100 watts into the home. By separating the power at the tap, voltage limiting devices in power passing taps will not be necessary as the voltage is passed on a siamese coaxial cable with a twisted-pair copper conductor. Once all the codes have been satisfied, it is time to examine the equipment that this 90 VAC will power. Most manufacturers will need to provide a new power module to facilitate the use of 90 VAC. While some of the models are capable of sustaining up to 94 VAC, they cannot deal with the peak voltages that accompany a ferroresonant waveform producing 90 VAC true RMS. These voltage peaks can theoretically approach 112 VAC. You either need to change the power modules in the amplifiers or select a powering system with a peak voltage below that of the module being driven. One key factor in selecting these modules is in the useful "rail -to-rail" capability of the module. When the peak voltage rises, the bottom rail must also follow after a moderate increase. Currently there is a useful range of 20 volts for design purposes with 60 VAC powering. When a ferroresonant device is utilized for 90 VAC, the rail moves up and the design window increases to 30 VAC. This is a moderate increase allowing a reduction in the overall number of power supplies until the current limitation problem peaks at 10 amps. With a controlled delivery source, the current power modules can be maximized to deliver a range of 48 to 50 VAC. And with the reduced current capability from a controlled source, either further distance or more power can be achieved.

Once the power modules have been selected, the equipment figures into the equation again. Most line gear is only capable of passing 10 amps true RMS. The passives and distribution equipment varies from 6 amps up to 7.5 amps depending on the vendor. While work is in process to increase these current capabilities, the selection of a power delivery system that minimizes current in the network will allow for longer runs where current is concerned. In reality, the design moves back to a current limitation before the full usefulness of the 90 VAC change can be realized.

Finally, safety issues must be addressed. The only aspect of 90 VAC powering that everyone can agree upon when it comes to safety is that training will be necessary. However, with the reliability demands of Networks in the 90's, great care will be exercised not to casually open the network path. Modules cannot be swapped unless the system they are deployed in is capable of "Hot Swapping". If a unit is to be removed, there must be an identical unit or system to take its place with no break in the network traffic. This type of training is long overdue in the cable industry.

The effective result of utilizing a 90 volt system is a reduction in the number of power supplies. Current designs with 750 MHz or 1 GHz spacing require power supplies at every 0.6 to 1.2 miles depending on the geographic area. One system completed an upgrade where they utilized twice as many power supplies as they had previously. However, when the first utility bill arrived, the total dollar amount was closer to triple of that which they had previously received. Obviously, the more efficient a system is at powering, the lower the operating costs for powering will be.

## Express Power Feeders

When you examine any powering design, the highest losses occur in the cable segments closest to the power source. If you can reduce the amount of current flowing in these segments, the voltage drop decreases proportionally. One technique for maximizing the ability of any power supply, 60 VAC or 90 VAC, is to run an "express feeder" from the power supply past the first load point. For example, if the voltage drop in a 3000' segment is 12 VAC and the load is 4 amps, by utilizing a coaxial extension cord, the voltage drop can reduce the load to 2 amps on each segment. This results in a uniform 6 VAC drop on each leg. This would allow another section to be installed, again increasing the voltage drop on the express feeder to 12 VAC. This is a proportional gain that allows approximately only 40% of the segment distance to be gained. But sometimes that small amount is the difference between the need for an additional power supply.

Typically, .825 coax is utilized due to its ability to carry high current loads with nominal loop resistance. Different sizes of cable can be utilized depending on the desired end voltage or current load of the design. This technique could also allow a leg in a single direction to have a total current draw of 12 amps with the current splitting 3 amps to the first segment and 9 amps to the remaining plant. Typically there is no RF transmission on these express feeders. Typical gains in utilizing this technique are a 10-20% decrease in power supplies in the system or better. Again, the geographic layout of the system will determine the total benefit. The down side of this technique is in direct burial applications or where conduit systems are already full. The additional cost in these locations merits close examination of the costs. Don't forget the operating costs that will be attributable to another supply location such as fixed clerical fees or attachment fees.

## Architectures

There are many system designs being deployed today for communication networks. With regards to powering, the only common thread seems to be the centralization of powering. There is a desire to locate the power source at a central area and deliver power to the system. Some designs will locate the unit at the fiber node. If a node size is large enough, it may be the only one connected to that power supply. Other designers will utilize smaller nodes and then interconnect them utilizing express feeders and 90 VAC systems. Yet other designers are looking for techniques to harden the network against the ever popular "Semi Fade". That's where a "Mack" truck takes out a line or an entire power supply. To accomplish this task, they want to utilize "smart" switches that will seek the first active line. When the primary line fails, the next active line comes up to feed the fiber node. This process must be accomplished in less than the hold-up time of the fiber node. Even when we strive to achieve near passive system designs, the powering of home devices and the possibility of powering "PCS" gear requires that we still reliably power the network.

## Efficient Delivery

To address the capacitive nature of today's networks, a new form of delivery was needed. The **Unity Wave™** was designed to solve this problem as well as the problem of transmission efficiency and $I^2R$ losses. A *trapezoidal waveform* proved to be the solution. The waveform has a controllable rise time which controls harmonic content for reduced hum modulation in the network elements. This waveform lowers the overall RMS current being required by the cable network by increasing the load power factor. This in turn *lowers the cable segment voltage losses* allowing a higher voltage at the end of line or a deeper penetration into the cable network. Also, the $I^2R$ losses in each cable segment are minimized *lowering the overall system power requirements* and maximizing transmission efficiency. The flat crest or peak voltage allows the switchmode power packs to draw their current over a longer duration which is the mechanism for power factor improvement. An additional benefit is that the peak voltage of the output is virtually equal to the RMS voltage. A **Unity Wave™** product with a 90 VAC RMS output will have only a 91.5 VAC peak voltage. Standard CATV power supplies can have a peak voltage up to 40% higher than their RMS output.

The final improvement is the output frequency. A **Unity Wave™** product has an *adjustable output frequency.* DC to 60 Hz can be selected by programming the unit in the field through its status monitoring or network management system. Personnel safety is improved at frequencies below 5 Hz, but corrosion performance improves as the frequency increases. The **Unity Wave™** product permits the output frequency to be optimized to suit each customers requirements.

## Higher Delivery Voltage and Current Capability for Increased Penetration

The need to minimize the number of locations for maintenance purposes first became clear when evaluating the maintenance of the batteries required to provide 8 hours of backup in a 500 home node. It is estimated that it would take from 4 to eight power supplies with 6 batteries each to provide the requirements set forward. This translates into 48 batteries every 3.5 years at a material cost of $4224.00. In addition, the human resources needed to maintain these batteries during their usable life overshadows the material cost. The solution was to create a product that *centralized the standby operations on a per node basis.* To facilitate this, a higher output voltage is required.

Each leg of the system is limited today to 10 amps RMS maximum current by the coaxial devices employed in the system. The *lower peak currents of the trapezoidal waveform allow more power to be delivered into the coax.* If only 60 VAC RMS is used, the maximum power per feeder would be 600 watts. By utilizing 90 VAC RMS, the maximum is increased to 900 watts per feeder. With a system design of 4 feeders per node, the power output available is 3600 watts into the feeders. This is a substantial gain over power supplies currently used in CATV systems.

Most of the line gear manufacturers are producing equipment that will operate on a "controlled waveform" 90 VAC RMS supply or are in the design stages of such a product. A "controlled waveform" refers to the fact that the peak voltage of the **Unity Wave™** products is only 2% greater than the RMS output voltage.

## Increased Reliability Through Redundancy

An optional feature of the **Unity Wave™** power supply is its *"N+1" redundancy.* This means that no single failed module will prevent operation of the network. This will in turn signal the Network Management system to notify system personnel of the problem. The Network Management system can control and monitor numerous parameters of the system's operation.

## Extended Standby Operation

When faced with the large number of batteries required to provide 8 hours of emergency service, it is apparent that there must be a better solution. The **Unity Wave™** system employs an optional *DC Generator* to provide long term uninterrupted backup at a lower cost than batteries. The generator can be fed from either bottled liquid propane or a natural gas hookup. It is designed into a *cabinet which may be located separately from the power supply* location. The **Unity Wave™** system will run from internal battery power for 20 minutes before switching the generator on. This will handle most nuisance outages. If a DC generator is not practical in a given area, a system utilizing additional batteries could be deployed for a total of eight hours of backup time or greater.

# TWO-WAY PLANT CHARACTERIZATION

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## ABSTRACT

*An array of new telecommunication services that encompasses video, voice, and data offers the cable television industry a tantalizing opportunity that can be capitalized by taking advantage of the broadband nature of the CATV networks. Although most cable systems can be upgraded to full bi-directional capability, the amount of experience and knowledge on the upstream or the reverse portion of the networks are inadequate relative to those on the traditional, forward direction. This paper describes two-way test procedures, equipment package, and field test results and experiences obtained from five cable testbeds.*

## INTRODUCTION

The broadband nature of the CATV networks fortified with fiber deployment and digital technologies uniquely position the cable industry as a full or universal telecommunications provider. The next generation cable network architecture envisioned by Cable Television Laboratories[1] describes a regional hub interconnecting groups of fiber headends, which in turn feed fiber hubs, and finally to fiber nodes. Each fiber node serves about 200 homes passed which are connected via existing coaxial tree-and-branch network.

The typical coaxial reverse bandwidth of 5 to 30 MHz is considered by CableLabs to be only an interim solution until other methods are made available such as mid-split, passive, and parallel coaxial networks. However, the existing sub-low reverse network may continue to thrive for an extended period of time, and this network asset is the critical element in realizing the full potential of the cable network. Consequently, the reverse portion's noise characteristic and transmission capability must be better understood and controlled if the myriad of new services are to be successfully deployed in the absence of fully segmented, digital cable network.

In order to better study this problem, the CableLabs Telecommunications Subcommittee commissioned a subset of its members to participate on the Network Integrity/Ingress Working Group. The participants of the working group were Mike Stone and Ray Fournier of Continental Cablevision, George Hart and Albert Kim of Rogers Engineering, Larry Langevin of Greater Media, Bill Gast of Crown Cable, and George Stickler and Paul Schauer of Jones Intercable. The working group was chaired by Paul Schauer. The purpose of this working group was to direct a two-way field test measurement and analysis program. The main objectives of the test program were to:

- characterize the two-way CATV plant analog impairments;
- determine the impact of these impairments on the performance of the digital transmission;
- recommend guidelines to the member companies of CableLabs as to appropriate network topologies and operations methods to guarantee reliable digital transmission.

To support the working group's efforts in meeting the test program objectives, the services of three separate companies were contracted by CableLabs. These companies were AT&T Bell Laboratories, GT Communications, and Rogers Engineering. AT&T contributed to the digital portion of the test platform, Rogers Engineering consolidated the digital portion with the analog testing segment and developed the automated test equipment control and data collection software, and GT Communications retrieved the field data and performed preliminary analysis.

The first section of the paper describes the two-way test procedures, test platform, and data analysis methodology. The second portion provides the testbed system details and field test results from the five test locations. Finally, lessons learned from the field efforts are discussed to highlight the necessary steps required to attaining reliable two-way plant.

## TEST PROCEDURES

The basic premise of this undertaking was to ascertain what CATV system elements affect digital transmission and how they can be managed or controlled given a "generic" digital communication system. The digital transmission system utilized in the test was a QPSK modulated, T1 rate channel with no error correction. The intent was not to derive the necessary modulation technique and error correction scheme to properly operate in a given CATV system; rather the intent was to determine the principal sources of impairments existing in CATV plants and educe corrective measures. Following this axiom, the two-way plant characterization effort was divided into three broad areas:

1. collect valid test data over sufficient time period to properly characterize the baseline two-way digital performance for each test system ("valid" in the sense that the proposed test equipment package is set up properly and the procedures are followed correctly; "sufficient" period in the sense that enough samples are collected to derive statistically valid analysis);
2. identify primary symptoms that produce digital transmission impairments, infer potential causes of the identified symptoms, and conduct controlled plant variations to ascertain the casual agents;
3. derive a set of recommendations or guidelines to achieve optimum CATV plant and digital equipment configuration and desired signal quality and reliability.

The analog and digital testing were conducted in parallel to permit correlation of digital error occurrences to the changes in the analog parameters such as ingress and impulse noise levels. Other factors contributing to digital error performance and availability were also accounted for to supplement the correlation effort such as cable maintenance practices, plant equipment failures, power outages, and environmental conditions. The analog and digital parameters comprised the core set of data and were measured and recorded by the control software program automatically.

The core analog and digital data measurement technique defined in the two-way test procedures[2] took a broad perspective on the cause-and-effect relationships between different cable systems and the digital performance. The classical transfer function definition of the cable channel was not addressed with these test procedures. Efforts to define the cable channel characteristics to such detail has been undertaken by CableLabs to evaluate complex modulation techniques for the transmission of compressed digital video information[3]. Results from this experiment will yield valuable insights for modem designers in developing appropriate cable terminal equipment.

A comprehensive and consistent set of data is a necessary prerequisite to draw valid and useful conclusions and recommendations. The test procedures enable a consistent collection of field data across a variety of plant design, topology, operation, and environmental conditions. The consistent plant characterization methodology provides a foundation upon which to compare and assess the effectiveness of various plant improvement techniques in enhancing the cable plant transparency to digital transmission. Such techniques will attain wide adaptability and high confidence level if all test participants realize similar improvements. Hence, the test procedures provide a powerful method of vetting improvement measures and thus enhancing the utility of the overall test efforts. The conclusions reached through this process will lead to guidelines for cable operators wishing to implement digital data services on their cable networks.

## Analog Performance Parameters

Prior to any digital transmission tests, the forward and reverse cable plant analog parameters need to be recorded. The parameters are divided into static and dynamic categories, and the latter category has two subdivisions between fast-varying and slow-varying types. The parameters under the static category are not expected to vary significantly over time, and include the following:

- amplitude-frequency response;
- group delay response;
- visual carrier-to-noise (CNR);
- distortions (CTB, CSO);
- hum modulation;
- return loss.

Under the dynamic category, ingress and impulse noise comprise the slow-varying and fast-varying parameters, respectively. Ingress is the level of unwanted ambient signals leaking into the cable plant, and is a direct result of imperfect shielding of the cable plant. Ingress mechanisms include loose connectors, cracked cables, broken multitap spigots, and other plant defects. On the reverse plant, sources of off-air ingress include short-wave broadcasts, HAM band transmissions, and CB traffic. The short-wave broadcasts originate from distant transmitters, thus subjecting the cable plant to a uniform field. In contrast, the HAM and CB transmissions are localized and will affect a small portion of the plant near these HF energy origination.

The ingress spectral contents are confined to narrow frequency ranges; however, the impulse noise energy occupies a continuum of frequency spectrum with decreasing spectral energy density as frequency increases. Impulse noise occurrences are sporadic and their causes are not well understood. However, these broadband noise bursts have been linked to electrical discharges (AC arcing) which materialize when intermittent connections along the transmission path are encountered such as improperly installed or deteriorated connectors and splices, and cracked/eroded cables. Other common causes of impulse noise are lightening, powerline arcing, arc welding, and industrial machinery.

The test participants should ensure that their cable testbeds meet minimum analog performance requirements. For the forward direction, the minimum performance is tied to the FCC Rules and Regulations document, Part 76, as amended through the Cable Act of 1992. However, no equivalent industry standards or regulatory specifications exist for the reverse plant static parameter performance and for the dynamic variables. The two-way test efforts will attempt to fill this gap by measuring the intensity and incidence rate of ingress and impulse noise for various plant configurations. These measurements will define the baseline performance of a given test site. All analog parameters are measured with HP 8590/91E Spectrum Analyzer.

This initial cursory examination of the cable plant for two-way testing may uncover the need for repairs and realignment. Proper plant alignment according to individual operator's design guidelines must be verified prior to digital testing. To ascertain plant shielding integrity, CLI radiation patrols or CB "sweeps" can be performed. However, the repair activities should be confined to what each operator considers as "normal maintenance" for its plant. Disproportionate amount of repairs before conducting the digital testing may lead to misleading results which do not correspond to a "typical" CATV plant. Various improvement techniques will be exercised in a controlled manner under the direction of the working group after initial or typical digital performance is recorded for each test site.

## Digital Performance Parameters

The basic metric used to determine digital link performance is bit errors or bit error ratio (BER). The telecommunication industry characterizes the digital transmission performance on the basis of statistical information derived from received bit error counts and received bit counts at 1-second intervals. Additional digital performance parameters are defined based on the bit error counts:

- availability;
- errored seconds (ES);
- severely errored seconds (SES);
- degraded minutes (DM).

There are numerous standards available from which to measure and characterize the digital link performance such as ANSI, AT&T Accunet T1.5, and CCITT Recommendation G.821. Each standard allocates different performance requirements to the digital parameters listed above. One of the most widely used digital performance measure is the CCITT G.821 recommended performance objectives. The two-way test procedures include HP37701B T1/Datacom Tester to measure and record G.821-compatible digital parameters.

The G.821 recommended error performance objectives are defined for an end-to-end transmission encompassing the low, medium and high grades, and each grade is allotted a proportion of the total end-to-end objectives. The digital services carried on the cable plant should at a minimum meet the recommended performance for local exchange carrier (LEC) access which constitutes low grade performance. Connection between LEC and an interexchange carrier (IXC) constitutes medium grade performance, and between IXCs as high grade. The two-way testing will determine if this objective can be achieved with existing cable plants, and establish the effectiveness of various plant improvement techniques in meeting higher digital performance objectives. Exhibits 1 and 2 list the G.821 end-to-end error performance objectives and the apportionment among the three grade levels.

| Classification | Objectives |
|---|---|
| Errored Seconds | Fewer than 8% of one-second intervals to have any errors |
| Severely Errored Seconds | Fewer than 0.2% of one-second intervals to have a BER worse than $10^{-3}$ |
| Degraded Minutes | Fewer than 10% of one-minute intervals to have a BER worse than $10^{-6}$ |

**Exhibit 1: G.821 End-to-End Objectives**

| Grade | ES (%) | SES (%) | DM (%) |
|---|---|---|---|
| Local | 1.2 (X2) | 0.015 (X2) | 1.5 (X2) |
| Medium | 1.2 (X2) | 0.015 (X2) | 1.5 (X2) |
| High | 3.2 | 0.04 | 4.0 |

**Exhibit 2: G.821 Apportionment**

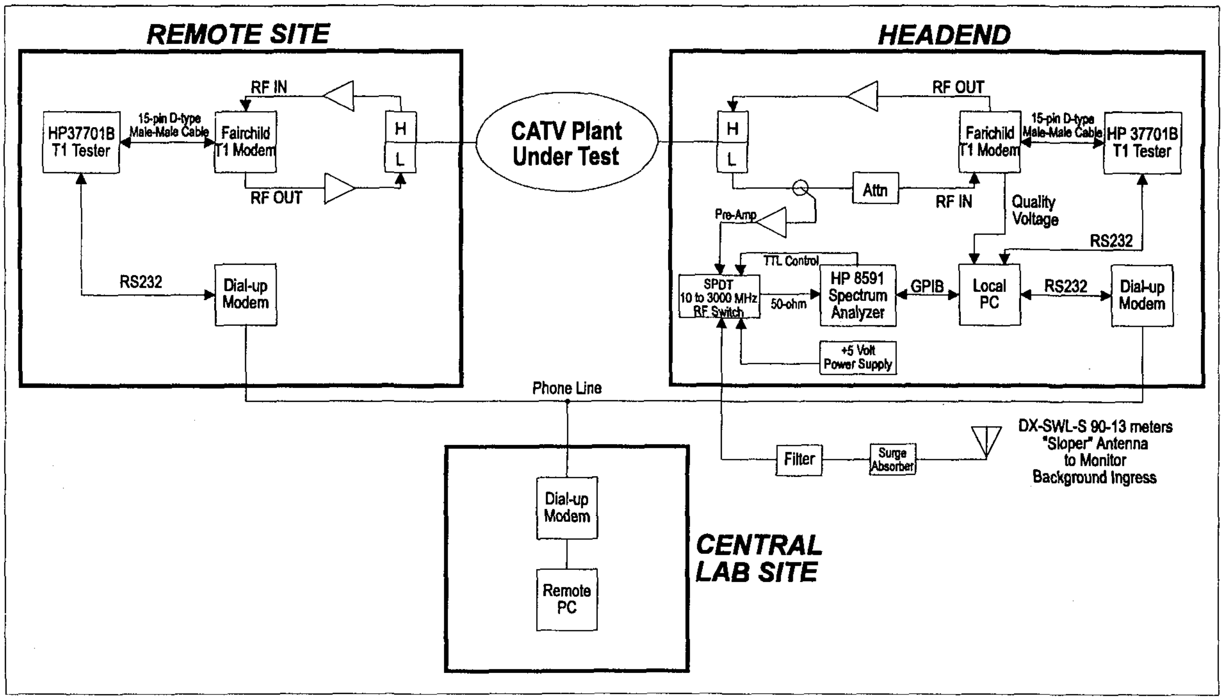



**Exhibit 3: Two-Way Test Platform**

## TEST PLATFORM

Exhibit 3 illustrates the two-way test platform. The equipment control software (written in QBASIC) residing on an IBM-PC compatible computer at the headend manages and directs the activities of two principal test equipment: the spectrum analyzer and the T1 tester. Another element controlled by the program is an RF switch to select between the local HF antenna feed and the reverse frequency spectrum feed from the testbed. The T1 tester at the remote site is not controlled by the QBASIC program; instead, the forward T1 tester is controlled remotely using a communications software package (Procomm Plus).

Two T1 testers are used, one at the headend or hub and the other located at the remote site to measure the reverse and forward digital performances, respectively. The T1 testers generate 1.544 Mbps pseudo random T1 data streams and perform data signal analysis as per CCITT Recommended G.821. Two Fairchild M505 Broadband Modems modulate the T1 streams in QPSK format onto RF carriers: between 5 to 42 MHz for the reverse path and between 50 to 400 MHz for the forward direction. No error correction mechanism is utilized by the modems. The T1 testers are configured to record error activities and alarm conditions in 1-minute samples. At the end of a test run, the tester provides the summary digital performance (BER, ES, SES, DM, and availability) and the 1-minute resolution sample results.

The QBASIC program collects several different types of raw data from the spectrum analyzer in both frequency-domain and time-domain. Changes to the method of collecting and measuring data or test procedures are implemented easily by downloading a modified QBASIC program to the appropriate test site computer. Since the program is written in Basic and operates on a PC, the test software is easy to use and maintain. The program has operated effectively in the beta test sites for over several months which is indicative of the stability of the test hardware and software. Explanations for each raw data type and measurement methods are described below.

### HF Antenna & Ingress Files

The antenna trace file captures the ambient high frequency band energy activity; the reverse band frequency trace file captures the ingress and noise levels present in the reverse portion of the cable plant. Both traces are swept from 5 to 35 MHz. The antenna and ingress traces are created every 30 minutes, one immediately after the other to allow for time correlation. The antenna/ingress subroutine sets the spectrum analyzer instrumentation to RBW of 100 kHz and VBW of 300 Hz. The resulting sweep time is 3.0 seconds.

The ingress monitoring allows full visibility over the entire sub-low band, revealing amount of useable frequencies (either in slots or contiguous), availability (depending on service definition), and carrier level required to meet specified service quality and reliability. Because the reverse carrier level has a limitation due to equipment and design, the intensity and incidence of ingress levels observed broadly highlights the level of effort required to improve the plant integrity before a variety of services can be supported reliably. In addition, the ingress monitoring routine also provides an account of the plant's transmission integrity and problems such as amplifier failures, power supply failures, amplifier by-pass, and signal interruptions (voluntary or involuntary). The measured ingress and noise levels on the reverse plant are processed with respect to frequency, percentiles (deciles, quartiles, median), and time (diurnal). The profile of ingress activity is cross-correlated to the following:

1. plant size;
2. number and type of subscribers served (SFU, MDU);
3. construction type (aerial, buried).

Additional correlation to specific plant portions (trunk, distribution, drop, home wiring) can be conducted by focusing on one specific bridger area.

The antenna monitoring, in conjunction with the ingress monitoring, provides a reference base to compare relative shielding effectiveness of the testbed. The shielding effectiveness derived will not be in absolute terms since the HF antenna employed is not calibrated, but the ambient levels measured can be used to calculate the relative shielding effectiveness of a specific testbed germane to different serving sizes, maintenance activities, and repair efforts.

**Error-Triggered Power Time Sweep Files**

The error-triggered power time sweep file (TR file) captures the interference energy that has caused bit errors on the digital carrier. The error occurrences are appraised via a quality voltage signal level available from a test point on the receive section of the Fairchild M505 T1 Broadband Modem, and the voltage level is used to trigger the power time sweep subroutine. The DC quality voltage provides an indication of the amount of noise on the demodulated in-phase (I) data carrier. The voltage level is sampled through a 12-bit A/D card residing on the local computer. When this voltage exceeds a preset threshold level saved in the subroutine, the current time sweep trace is captured in a file. The spectrum analyzer is set to zero-span mode with RBW of 1 MHz, VBW of 1 MHz, and sweep time of 100 ms for the power time sweep traces.

Regardless of where in the sweep the threshold violation trigger is detected, the current sweep is commanded to complete the 100 ms trace. Once the sweep is completed, the entire 100 ms trace information is uploaded into a file consisting of date, time, threshold preset level, quality voltage level, and the 401 data points of the spectrum analyzer trace (fixed resolution of the HP8591). The zero-span center frequency is selected ± 3 MHz away from the digital carrier frequency. This particular frequency offset is chosen to be close enough to the digital carrier's skirt to have visibility on the carrier but at the same time far enough away to have sufficient dynamic range in capturing the interference energy.

One disadvantage with this method arises from the indirect measurement technique—the actual interference energy that causes the bit errors may not always be captured because visibility underneath the digital carrier is not possible with this method. However, by placing the reverse digital carrier at an ingress-free spectrum, there is high probability that majority of interference causing logic errors is due to impulse noise. The choice of analyzer settings and location of the center frequency of the zero-span provide visibility on the short-duration impulse activity in terms of incidence and intensity. There are recognizable energy patterns captured by the traces and thus can be catalogued according to severity of error hits, coarse causes of interference, duration, plant construction, and plant size. Unfortunately, processing the impulse noise by energy patterns is time-consuming and does not answer what the root causes of the interference are. Nonetheless, the traces do provide indirect clues as to where to look for the causes of impulse noise interference.

**Reverse and Forward Digital Files**

The reverse and forward digital files provide all the information related to the digital transmission performance during the test run. The digital files contain the summary digital results, the 1 minute resolution digital records, and a host of alarm flags.

The forward digital carrier level is set -20 dBc relative to the adjacent video carrier level, and the frequency is set at 1.5 MHz below the adjacent video frequency. Similarly, the reverse digital carrier level is set -10 dBc relative to reverse video carrier level as specified by the design of the return plant. Note that not all test participants have followed the -10 dBc setup on the reverse due to noise and ingress levels present at the associated test site. Some carriers are set at reverse video level or higher. The carrier level is adjusted with attenuators such that its input to the spectrum analyzer is at a fixed level of 10 dBmV (in 1 MHz RBW). The frequency of the reverse carrier varies between test participants, ranging between 17.0 MHz to 26.5 MHz.

The digital performance measurement is the fundamental metric in the two-way characterization effort. The digital results provide a direct visibility on impulse noise activity, plant maintenance, interruptions, and power failures. By placing the reverse digital carrier at known ingress-free frequency ("clear" spectrum), the causes of errors can be confined to those causes listed above. The HP37701B T1/Datacom Tester is configured to record digital performance information in 1 minute increments, providing a suitable resolution of error and alarm occurrences.

The individual 1 minute records are processed into BER histogram and into error occurrence in time. The former process provides the distribution of errors in the data set and the latter shows when errors occurred in time. Both processed results are used to isolate or highlight large error occurrences in order to conduct further investigation with the local operators in determining possible causes. The forward and reverse error occurrences are compared in time to ascertain if correlation exists.

The summary and individual 1 minute records furnish a means to gauge the behavior of error occurrences: low logic error counts arising steadily versus very large error counts but only sporadically. The ingress and error-triggered traces supply additional, supporting intelligence in analyzing the digital error performance and behavior.

## TEST SITE DESCRIPTION

Five test participants were involved in the two-way plant characterization effort: four beta testbeds (Site A to D) and one alpha testbed (Site E). Pertinent testbed information for each site is summarized in Exhibit 4. The testbeds encompass a wide range with regards to number of subscribers served, types of subscribers served, number of active elements, plant construction, and geographical dispersion (northeast, midwest, southeast continental US). It is worth noting that all test participants have had several years of experience with reverse plant operations, notably in supporting impulse pay-per-view (IPPV) services and status monitoring communications systems.

Site A testbed is the smallest test area with respect to number of subscribers fed, serving a total of 1329 subscribers of which 94% of them are multiple dwelling unit (MDU) customers. Majority of the MDU buildings are low-rise complexes. Approximately 25% of the customers in the test area subscribe to IPPV service, and the reverse IPPV carrier is located at 8.9 MHz. Site A also has status monitoring (SM) carrier at 29.5 MHz.

| Parameters | Site A | Site B | Site C | Site D | Site E |
|---|---|---|---|---|---|
| No. of Subscribers SFU<br>MDU | 83<br>1,246 | 3,821<br>472 | 1,700<br>2,000 | 2,000<br>1,000 | 4,962<br>2,987 |
| Fiber Link Length (mile) | 6.95 | 3.0 | 5.27 | 3.0 | All Coax |
| Return Laser Type | Jerrold | ALS | Texscan | Scientific Atlanta | N/A |
| Trunk Plant (mile) | 1.48 | 23.0 | 30.0 | 3.0 | 17.4 |
| No. of Trunk Amps. | 8 | 55 | 56 | 9 | 87 |
| Maximum Trunk Cascade | 5 | 9 | 13 | 9 | 25 |
| Digital Modem Cascade | 3 | 9 | 5 | 9 | 25 |
| No. of Distribution Amps | 32 | 178 | 156 | 1 | 396 |
| Maximum Dist. Cascade | 2 | 2 | 2 | 1 | 3 |
| % Aerial | 63.1 | 75 | 97 | 10 | 50 |
| % Buried | 36.9 | 25 | 3 | 90 | 50 |
| Bridger Switching Support | Yes | No | No | N/A | Yes |
| Return Plant Applications | SM @ 29.5 MHz<br>IPPV @ 8.9 MHz | IPPV @ 8.5 MHz | IPPV @ 11 MHz<br>Modem @ 6.2 MHz | None | SM @ 18.0 MHz |
| Home Terminal Equipment | Jerrold | Jerrold | Zenith | N/A | Zenith |
| Reverse Video Level | 25 dBmV | 20 dBmV | 17 dBmV | 20 dBmV | 20 dBmV |
| Reverse Digital Level | 15 dBmV | 10 dBmV | 19 dBmV | 10 dBmV | 20 dBmV |
| Average Plant Age | 7 years | 12 years | 5 years | 10 years | 15 years |

**Exhibit 4: Testbed System Information**

Site B testbed is the second largest test site with a total of 4293 subscribers of which 89% of them are single family units (SFU), and majority of the plant is aerial. Similar to Site A, the reverse spectrum includes an IPPV carrier at 8.5 MHz; no other services on the reverse spectrum is carried. The reverse digital carrier was carried at 10 dBmV referenced to the reverse trunk input, which is 5 dB lower than that of Site A. This particular cable system utilizes HRC channel lineup in the forward direction.

Site C testbed size is similar to that of Site B in terms of number of subscribers, trunk amplifiers and line extenders. Total number of subscribers served in the testbed is 3700 of which 54% of them is MDU. This system utilizes 11.0 MHz for its IPPV service and a modem carrier at 6.2 MHz. The reverse digital carrier is injected into the plant 2 dB higher than the reverse trunk input level of 17 dBmV instead of being -10 dBc. Higher level was required to overcome noise present on the sub-low and to acquire modem synchronization, which was surprising in light of the relatively new plant.

Site D's testbed has the lowest number of active distribution amplifiers with only one line extender conducting sub-low energy along with the plant portion between the trunk bridger ports and the first line extenders. However, the area being served is very dense with a total of about 3000 subscribers. Almost all of the testbed's plant is buried.

Site E's testbed is the largest test system with close to 90 trunk amplifiers and almost 400 line extenders. The testbed serves nearly 8000 subscribers of which 62% of them are SFU. The plant construction is evenly distributed between aerial and buried types. There is no return fiber optic link included in the reverse transmission path. Note that the reverse digital carrier was injected into the plant at reverse video trunk input level of 20 dBmV (0 dBc). Moreover, the average age of the plant was the oldest, with some portions of the plant exceeding 25-plus years.

Another distinguishing aspect of the Site E testbed was the utilization of the remote bridger switching to alter the testbed configuration. The bridger switching is separated into two bands: 5-18 and 21-33 MHz (18-21 MHz is the cross-over band) which can be exercised independently, and status monitoring system carrier operates at 18 MHz. Seven different configurations were exercised at Site E of which five of them are described in Exhibit 5 (designated CONFIG 1 to 5); the other two configurations are "Trunk Only" and "All On". The reverse energy contribution from various distribution areas were controlled via the bridger switched to artificially alter the test plant; however, the entire trunk portion of the testbed is reverse active for all test configurations. The number of trunk amplifiers shown in Exhibit 5 indicates the number of bridger switches that were activated to conduct reverse energy from the distribution plant.

When a trunk transponder is polled, its 5-18 MHz band switch is activated temporarily to communicate, and the polling takes place continuously. Therefore, ingress level measurements in the 5-18 MHz band for the six test configurations (excluding the "All On" case) include some energy from plant areas outside the specified configuration. However, the duration of the switch activation is very short and do not materially affect the results.

| System Variables | CON-FIG 1 | CON-FIG 2 | CON-FIG 3 | CON-FIG 4 | CON-FIG 5 |
|---|---|---|---|---|---|
| #SFU | 751 | 1032 | 1426 | 1489 | 1296 |
| #MDU | 0 | 55 | 0 | 55 | 2932 |
| # Trunks | 18 | 15 | 25 | 50 | 19 |
| % Aerial | 16.7 | 70.4 | 34.4 | 65.4 | 75 |
| % Buried | 83.3 | 29.6 | 65.6 | 34.6 | 25 |

**Exhibit 5: Site E Test Configuration Parameters**

## TEST RESULTS & ANALYSIS

The field testing was divided into three phases: first, an initial two-way measurement was conducted at the alpha site (Site E) to confirm the proper operation of the test platform and programs, and to make any required modifications; second, baseline performance results were obtained from all five testbeds, and principal symptoms for each testbeds were diagnosed; and third, controlled variation testing was conducted to substantiate the postulated casual agents identified in the second phase. Test results and analysis of the baseline and controlled variations testing phases are discussed below.

### Baseline Performance Results

After completing preliminary testing at Site E, the refined test platform was installed at the four beta test locations (Site A to D). The data and analysis presented in this section is based on test results collected over at least 27 consecutive days for the four beta test sites and one week per test configuration at Site E. Moreover, the data sets were collected after all problems associated with the test equipment package were corrected and confirmed for proper operation; hence, these data sets constitute valid baseline performance measurements.

The summary results for the five participating cable operators are presented below under anonymous names. The digital summary results are presented along with examples of reverse spectrum energy levels, BER histograms, and error-triggered power traces. Exhibit 6 lists the baseline digital summary results for Sites A to D, and Exhibit 7 shows the results from Site E for seven different test configurations. An example of reverse ingress DQM (decile-quartile-median) level chart is depicted in Exhibit 8, Exhibit 9 presents the BER histograms for Site E, and examples of error-triggered traces are presented in Exhibit 10.

| Digital Summary Parameters | Site A | | Site B | | Site C | | Site D | |
|---|---|---|---|---|---|---|---|---|
| | Reverse | Forward | Reverse | Forward | Reverse | Forward | Reverse | Forward |
| Total Test Duration (minutes) | 36750 | 37811 | 26298 | 28621 | 22311 | 36966 | 22862 | 24493 |
| Weighted Average BER | 1.8E-5 | 1.7E-7 | 5.9E-5 | 3.3E-7 | 1.6E-5 | 2.7E-7 | 1.3E-6 | 4.9E-7 |
| Weighted Average %Avail. | 99.978 | 99.971 | 99.771 | 99.997 | 99.997 | 99.999 | 99.591 | 99.867 |
| Weighted Average %ES | 0.215 | 0 | 10.893 | 0.011 | 2.745 | 0.004 | 0.170 | 0.111 |
| Weighted Average %SES | 0.026 | 0 | 0.172 | 0.001 | 0.048 | 0.001 | 0.006 | 0.003 |
| Weighted Average %DM | 2.624 | 0 | 31.404 | 0.140 | 13.963 | 0.027 | 0.831 | 0.703 |
| Wght. Average %BER $\leq 10^{-6}$ | 97.329 | 99.971 | 63.930 | 99.856 | 84.247 | 99.971 | 98.754 | 99.162 |

**Exhibit 6: Sites A to D Baseline Summary Digital Performance Results**

| Digital Summary Parameters | Trunk Only | CONFIG 1 | CONFIG 2 | CONFIG 3 | CONFIG 4 | CONFIG 5 | All On | Forward Digital |
|---|---|---|---|---|---|---|---|---|
| Total Test Duration (minutes) | 11574 | 9203 | 8041 | 11887 | 8845 | 6712 | 6374 | 64269 |
| Weighted Average BER | 1.8E-6 | 4.5E-7 | 1.4E-7 | 4.7E-7 | 7.5E-7 | 2.2E-5 | 1.3E-4 | 1.4E-6 |
| Weighted Average %AVAIL | 100 | 100 | 100 | 97.911 | 100 | 99.867 | 97.929 | 100 |
| Weighted Average %ES | 3.557 | 1.560 | 0.853 | 1.150 | 1.821 | 11.724 | 14.812 | 0.048 |
| Weighted Average %SES | 0.002 | 0.005 | 0.001 | 0.002 | 0.003 | 0.072 | 0.155 | 0.001 |
| Weighted Average %DM | 6.169 | 5.101 | 2.301 | 3.266 | 6.004 | 38.601 | 41.444 | 0.187 |

**Exhibit 7: Site E Baseline Summary Digital Performance Results**

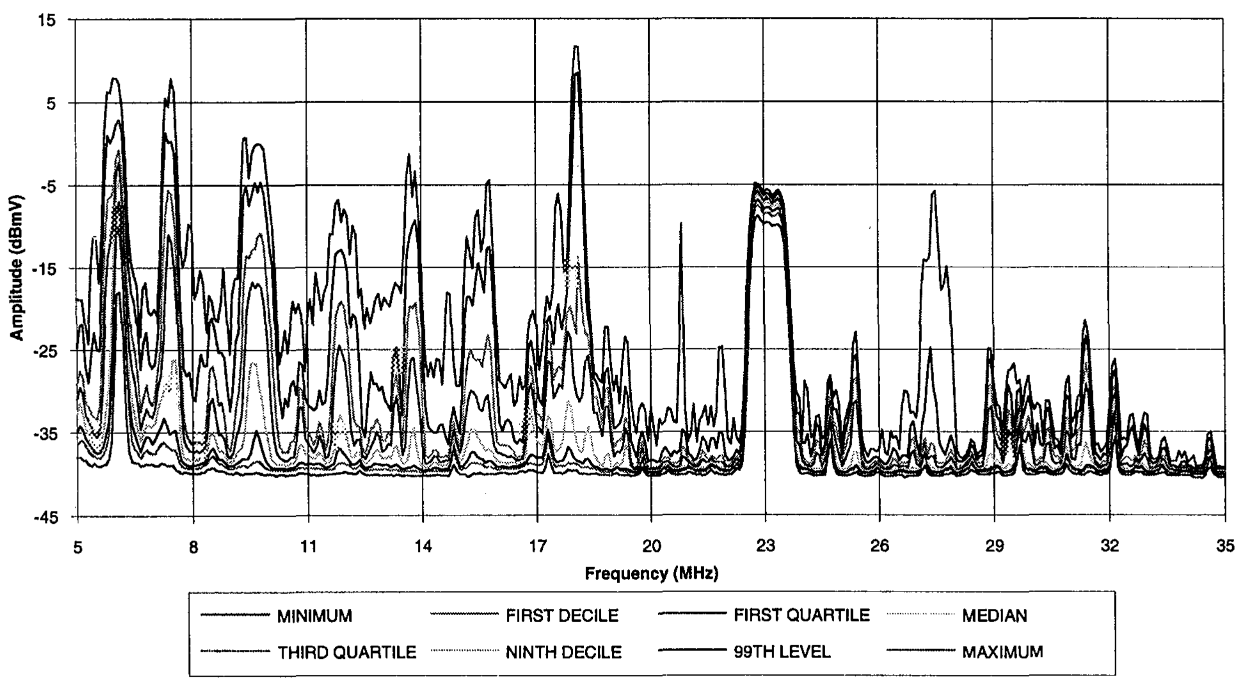



**Exhibit 8: Site E CONFIG 1 Consolidated Ingress DQM Levels**

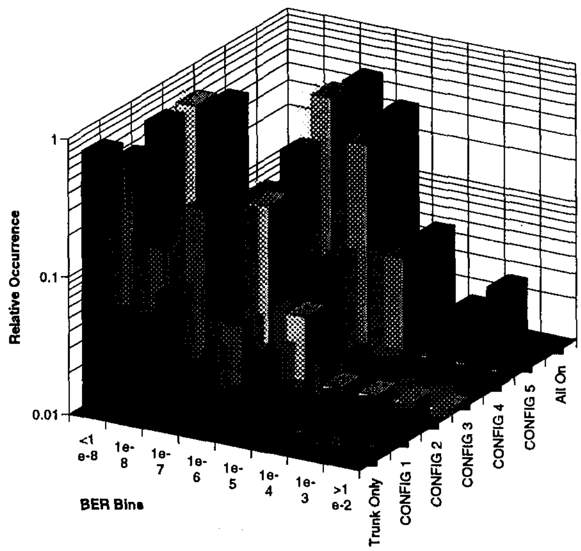
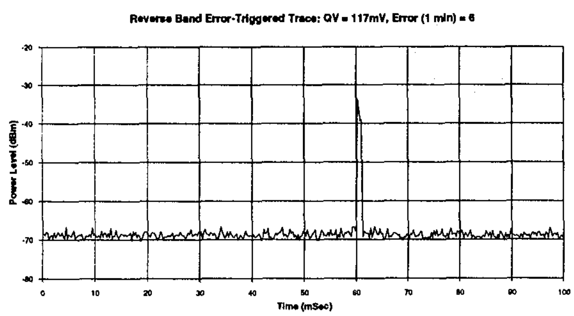
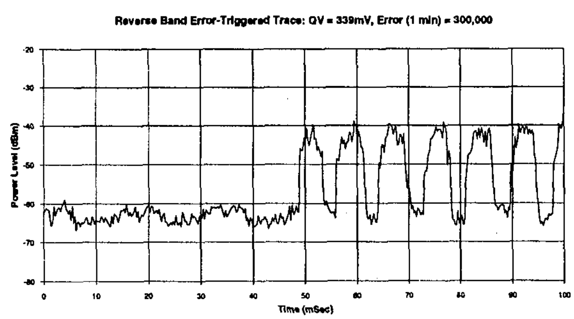
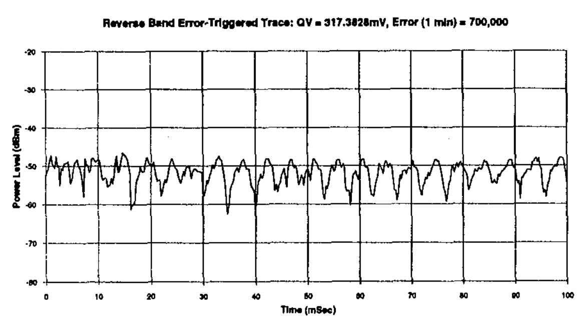
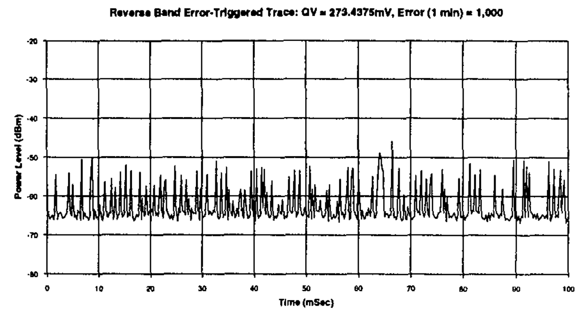



**Exhibit 9: Site E Summary Reverse Digital BER Histogram**



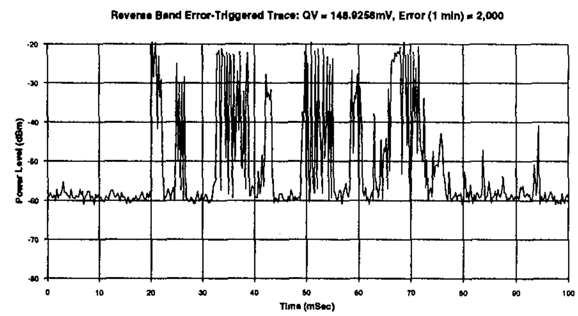




**Exhibit 10: Examples of Error-Triggered Traces**

## Diagnosis

The various types of test results collected during the baseline phase were consolidated to surmise sources of impairments. For some sites, the presence of poor weighted average BER in conjunction with low %ES and %SES suggested that errors occurred infrequently for short duration but inflicted high error counts. The occurrences of these large errors at very specific times with high regularity pointed toward some preprogrammed activity. In other sites, 60/120 Hz types of impulse noise dominated the digital performance, leading to suspect either internally or externally generated electrical problems. Improper reverse gain alignment and/or excessive reverse input levels were also witnessed, resulting in amplifier compression and/or laser clipping. In addition, the effects of weather—more specifically, thunder storms—were observed and correlated to digital impairments. Lastly, plant interruptions such as plant disconnections or amplifiers going into by-pass mode occurred very infrequently but their impact was substantial with respect to overall BER and availability.

## Controlled Variations

Based on the analysis and diagnosis performed on the baseline results, site-specific controlled variations were constructed to authenticate the postulated causal agents of impairments. Individualized controlled variations were particularly chosen to yield the greatest potential impact and improvement to the reverse digital performance. However, common variations for all test sites were implemented such as segmenting the testbeds to isolate and repair trouble spots.

Examples of site-specific controlled variation involved realigning the reverse plant, sweeping the reverse plant, deactivating existing reverse applications (IPPV carrier), selecting different IPPV carrier levels, selecting different reverse T1 digital carrier levels and frequencies, and installing high-pass filters at the tap. Each controlled variation test run constituted a 48-hour test duration. Several dozens of such test runs were conducted over a three-month period. Instead of showing the results of every single controlled variation test run, an overall discussion of the impact of the proactive changes and some hard lessons learned from the field trial are presented below.

## LESSONS LEARNED

### Fundamental Lessons

At a macro level, the size of the testbed certainly had a prominent role in determining the overall performance of the digital link. Strong correlations were found between the reverse digital measurements and the cable plant system parameters such as the number of trunk amplifiers and line extenders in a given testbed. Intuitively, such correlation was expected and the test data confirmed this supposition. The correlation also provided assurance that increasing plant segmentation will not only support increased system capacity but provide manageable network segments.

However, the very nature of the reverse plant, where all return signals share the same medium, does not guarantee a reliable reverse transmission just based on plant segmentation size. Only one plant defect can make the entire reverse spectrum unusable—in deed, several instances of field maintenance efforts confirmed this acute weakness of the CATV sub-low architecture where a single poorly aligned amplifier or loose/corroded connector adversely affected the reverse digital performance.

The meticulous care needed to maintain a reliable reverse plant with the combining and funneling effect was prominently highlighted during the course of the field testing. Maintaining proper operation for the forward channels alone was not a sufficient assurance for a dependable reverse spectrum operation. Many instances of asymmetrical impairments on the forward and reverse channels were witnessed; for example, impulse noise affecting the reverse band was effectively filtered out due to its decreasing power density with respect to frequency. In addition, the wide differential in digital link performance for the two directions from all testbeds further established fundamental differences of the forward and reverse links.

The numerous iterations of plant segmentations and plant maintenance efforts performed on the beta testbeds did not totally eliminate all sources of impairments—in some cases, reverse digital link performance could not be materially improved even after isolating and repairing some of the plant faults. However, such lack of success does not imply that the reverse sub-low operation is hopeless; rather, the field testing simply emphasized the critical importance of conducting a comprehensive and thorough plant hardening effort and maintaining precise gain alignment. **The necessity of accomplishing these two fundamental steps cannot be emphasized enough.** Although these steps seem too logical and self-evident to warrant such attention, the reality of the sub-low operation dictates such painstaking measures—in effect, there are no short cuts in acquiring and maintaining dependable reverse communication channel.

### OSS & Training

All levels of the organization must be involved and committed to realistically achieve a full two-way CATV communication network. The resources necessary to properly train the personnel and equipping them with requisite test equipment to deal with the sub-low spectrum must be expended in conjunction with the design and rebuild efforts that are primarily focused for the forward channel delivery. Nothing short of complete dedication will be needed to successfully achieve reliable reverse operations.

As stated previously, the increased penetration of fiber optic systems will mitigate the scope of reverse plant problems through smaller coaxial networks. But unless the segmentation process has reached down to very small networks, for example, each fiber receiver serving only tens of subscribers, many of the reverse plant problems witnessed during these field trials will continue to haunt cable operators. Having well trained and well equipped field personnel will be a key asset in sustaining reliable two-way plant; however, the value of possessing a comprehensive status monitoring and control system was also accentuated during the tests. Such troubleshooting aid was found to be very valuable and powerful in systematically characterizing individual bridger areas for isolating and uncovering faults.

Coupled with individually controllable reverse-band switches on every bridger amplifier, the network management system greatly improved the efficiency and effectiveness of field personnel by quickly isolating and directing problems for the technicians to handle. A comprehensive network management system will need to be adopted as network complexity increases in the migration of all-coaxial plants to a hybrid fiber-coax (HFC) architecture. Such adoption should include necessary switching and appropriate telemetry to monitor and control not only the forward plant but the reverse plant as well. The availability of remote monitoring and switching system for the reverse will be an essential tool in maintaining reliable interactive and transactional services.

The efforts expended to harden the reverse plant not only improved the shielding effectiveness by uncovering leakage points but also exposed potential transmission discontinuities that generated internal impulse noise, or AC arcing. Discontinuities such as loose/corroded connectors, splices and terminators; cracked cable sheath; water migration; poor grounding; and other plant defects severely affected the reverse digital performance. For some beta testbeds, the internally generated impulse noise was the dominate source of impairment, being much more harmful than ingress.

**Plant Integrity**

Efforts to track down the sources of impulse-generating plant defects proved to be very time-consuming as each and every connections and splices in the testbed had to be scrutinized. Cursory physical inspection did not always uncover the faults since the problems were intermittent. Increased impulse noise activity were observed with cold temperatures, high wind conditions, increased precipitation (including thunder storms), and combinations of all of these weather variables. The integrity of the transmission path was affected with adverse weather conditions, resulting in increased errored seconds, severely errored seconds, and poor availability.

Higher integrity seizing mechanisms and better weatherproof connectors and splices will certainly help in reducing ingress and impulse noise impairments. Although existing components do have good shielding and connection integrity, their performance limitations with respect to outdoor environment must be better understood in the reverse direction. The capacitance-coupling effect observed during the tests suggests that transmission integrity on the forward may mask impairments on the reverse direction, and that failure of transmission integrity occurs more quickly on the reverse direction than on the forward direction.

Along the same vein, the traditional method of measuring plant shielding effectiveness with leakage index did not successfully uncover specific problems affecting the reverse plant. The frequency of the leakage carrier does not accurately reflect the conditions of the sub-low band; a more appropriate and useful method is to use a CB radio to measure the level of leakage. Such testing must be done in accordance with governing radio transmission guidelines. Hence, much greater scrutiny must be applied to the overall plant transmission integrity than just on the forward direction.

**Drop Plant**

Some solace was provided with the observation of relatively clean trunk and distribution plants in comparison to the drops. Some of the controlled variation testing revealed that very little ingress was leaking within the hardline plants, which was not too surprising. Consequently, a thorough plant hardening exercise can be done once for the hardline portion and have sufficient assurance that this portion's integrity will be maintained for a certain length of time. Other advantages of the hardline plant are its accessibility and relatively limited number of connections. In contrast, the drop plant opens up a whole host of operational issues and poses a serious threat to the overall reverse plant viability. The drop plant possesses all the uncontrollable elements that cause nightmares for CATV operators: cables with poor shielding, difficulty in accessing connections, unknown illegal extensions, unterminated connections, poor connectors and splitters, improper installations, and micro-reflections.

Such uncontrolled additions to the CATV network were recognized in the past but were allowed to exist due to limited resources in tracking down all the culprits and their limited impact on the forward analog services. However, with the advent of digitally compressed signals, these unauthorized elements cannot be tolerated anymore. Equivalently, new services on the reverse plant cannot be successfully introduced unless the drop plant can be controlled and maintained at a much higher degree of discipline than is currently viable. For example, installing high-pass filters is not a viable long-term solution if equal-access services are to be provided.

New methods of consolidating and terminating the services offered by the CATV operator at outside/inside the home are being developed to address this issue, but technology alone will not alleviate the problems associated with the drops. New procedures and policies must be constructed for dealing with problematic subscribers and drop installations to regain control over the cable network. Tackling the drop plant may prove to be the most challenging and difficult of all problems mentioned so far. However, similar to the adaptation of network monitoring and control system with the evolution of HFC networks, the delivery of digitally compressed channels will provide the impetus in making the required changes in the drop plant which will be beneficial for the reverse services as well.

**Reverse Services**

After all the hard work is done to obtain a reliable and tight reverse plant, introducing ill-behaved reverse channels can totally jeopardize the entire sub-low spectrum. Such affliction was witnessed in couple of the beta testbeds involving the IPPV order system. The IPPV carrier only provided a narrowband service, several times a day, but the RF signature from the set-top boxes were so dirty that all of the reverse band was severely impaired when the carrier was activated. Understandably, the IPPV carrier was the only return signal in operation at some of the beta test sites prior to the two-way testing, and hence, the test participants did not have any visibility on its harmful effects.

However, the implications brought out by the IPPV effects were clear: that all services planned to be provided on the reverse spectrum must conform precisely to their technical performance specifications, and that intelligent control systems be developed to monitor individual carrier performances and to automatically shut down problematic carriers before they affect other services. Resembling the tight conformance of forward carriers, the reverse services also have to be maintained at a high degree of performance. Of course, the number of subscriber terminal equipment will far exceed the forward complement of equipment which will constrain the economic justifications for expensive, high performance home terminal devices. But as one beta participant summarized very succinctly, "You get what you pay for."

## CONCLUSIONS

A comprehensive two-way CATV plant characterization was undertaken under the auspices of CableLabs. The field test program jointly prepared by Rogers Engineering and AT&T Bell Laboratories was deployed successfully at five beta test sites. Valid and sufficient data was collected over a duration of approximately four weeks from all of the sites which represented the baseline performance levels. Principal symptoms for each testbed were diagnosed based on the analysis of the baseline test results, and a list of recommended controlled variations were derived. The baseline data was used as a yardstick in evaluating the controlled variations' impact on analog and digital performance.

After performing a considerable amount of field testing, many new insights into the CATV plant's viability of transporting digital services were obtained. The principal knowledge acquired from the testing efforts was the simple reality that the reverse sub-low spectrum is a very hostile environment. Although it's generally known that the reverse band suffers from the vagaries of HF broadcast, amateur, and citizen-band signal leakages, the two-way testing unveiled a much more complex and troublesome channel than previously recognized. In addition to ingress, impulse and common-path noise played significant roles in affecting the digital performance. In fact, ingress is benign compared to impulse noise; the former interference can be avoided with judicious selection of operating frequencies away from the known interferences, but it is difficult to avoid the latter type of impairment in which the entire reverse band can be affected.

The overall two-way CATV plant characterization effort represents the first step in examining the viability of the reverse sub-low spectrum in supporting digital services. The anticipated problems based on limited reverse plant experience and speculation were confirmed; in fact, the reverse plant proved to be much worse than expected in some respects, revealing a very hostile environment. The limited impact of ingress and the dominance of impulse noise on the digital link performance shed new light on the reverse spectrum, highlighting a different set of issues and problems that need to be resolved by potential modem designers. Of course, the burden of dealing with impulse noise must be carried by the system operators as well in reducing the incidence and intensity of broadband impairments.

Meticulous plant hardening and precise gain alignment efforts are required to obtain any semblance of long-term reverse transmission reliability—there are no short cuts. Such endeavor will not succeed without full commitment from all levels of the organization, and the necessary resources must be expended in providing the needed training, and providing requisite equipment and tools. However, these tangible, capital elements only comprise a subset of changes that are necessary; the attitudes of all organizational members must also be challenged and altered to properly reflect the changing definition of the CATV industry. The increasing competition and greater focus on customer satisfaction, customer service, and network reliability will compel such changes in attitudes to take place.


## ACKNOWLEDGMENT

The author wishes to thank CableLabs for its initiative and continued support; Network Ingress/ Integrity Working Group for its directions; beta test participants for their tremendous efforts; GT Communications and AT&T for their contributions; and colleagues at Rogers for their assistance and support of the two-way plant characterization project.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2. *Test Procedures for Two-Way CATV Plant Characterization*, prepared by Rogers Engineering and AT&T Bell Laboratories for CableLabs, October 1993
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# USER-PERCEIVED AVAILABILITY OF HYBRID FIBER/COAX NETWORKS

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## ABSTRACT

*Much attention has been focused recently on network reliability in both the telephone and cable television industries. CableLabs' Outage Reduction Task Force has formalized the modeling of distributions systems and suggested approximate component failure rates. Rogers Cable, in an extension of that work, analyzed several architectures and compared their performance.*

*This paper extends that work by predicting actual and perceived service availability to individual users, including the effects of drop and terminal equipment. The availability of a reference architecture is calculated, along with attainable improvements. Using the modifications suggested, the user-perceived availability of a video service is calculated to be nearly 0.9999, including the effects of headend, distribution system, drop cable and converter.*

## INTRODUCTION

The hybrid fiber/coax (HFC) distribution system has become the network of choice for both cable companies seeking to upgrade their facilities and for telephone companies seeking wider bandwidth and lower maintenance costs. While it is intuitively apparent that the reliability of HFC networks is much greater than long cascades of coaxial amplifiers in conventional trunk-feeder plants, it is important to quantify that improvement, as it must compare favorably with other alternatives for provision of both entertainment and non-entertainment services.

The Federal Communications Commission was so concerned about this issue (as well as the performance of other transmission alternatives), that it asked the Network Reliability Council to reconstitute itself for the primary purpose of studying the effect of new distribution technologies on the reliability of the nation's telephone system.[1] While wireless and satellite distribution will also be considered, HFC networks will be a primary area of emphasis because of its near universal planned deployment by multiple industries.

## HOW GOOD IS GOOD ENOUGH?

Before analyzing how reliable the new networks are, it is worthwhile setting realistic goals.

### Some Benchmarks

The widely quoted availability[2] of local telephone service is 99.99%, corresponding to a yearly outage time of about 53 minutes. That figure, however, does not include subscriber's telephones or in-house wiring (which is no longer the responsibility of the telephone company) and may not include the physical cable between homes and central switching offices.

A recent paper has proposed considerably more modest requirements for long-haul data circuits: 99.84% availability, with a predicted 99.96% availability of the local exchange carrier circuits on each end.[3]

Within the cable television industry, one of the few studies of viewer tolerance of failures was undertaken by CableLabs' Outage Reduction Task Force. Its principal report, *Outage Reduction*[4], was published in 1992 and included sections on customer expectations, methodology for tracking outages, reliability modeling methods, techniques for improving the reliability of network powering and system restoration techniques.

Based on a number of studies of customer reactions, CableLabs found that there was a sharp knee in customer perceptions of reliability at about 0.6 outages per month. In other words, customers who experienced less than about two outages every three months found the service to be acceptably reliable, while those experiencing more outages had a very negative opinion of the cable system reliability. Based on a mean time to repair (MTTR) of four hours, CableLabs translated 0.6 outages/month into a minimum acceptable availability of 99.7%.

Even given the different nature of the services provided over the public switched telephone network (PSTN) vs cable television systems, the difference in acceptable availability is very large. User perception of failures should be even greater.

User Perceptions of Availability

Standard telephone circuits are affected by many more outages than users perceive, both because of the low percentage of time the telephone is in use and because of how customers react to problems. For instance, if a call is interrupted because of an outage, the parties simply stop talking and one of them re-dials to re-establish the connection. Since the vast portion of the average telephone call is handled over shared facilities, there is a high probability that an alternate path will automatically be established by the network around any failed element. While the callers may be irritated by the call interruption, they are unlikely to identify the cause as an outage.

Since both parties are immediately aware of the failed connection, no loss of information occurs. Furthermore, since the average telephone user has the phone "off-hook" for less than a half a hour per day, outages occurring the other 23.5 hours do not affect perceived availability. Of course, as usage of the network for extended-duration data connections increases, subscriber are exposed to many more failures.

The situation for classical cable systems is radically different, both from network design and subscriber usage. In existing cable television systems, little if any of the network is redundant, so that circuits interrupted by failure cannot be re-established until the failed element is replaced. Furthermore, the desired programming that is not viewed generally represents irretrievably lost data (they don't stop the Superbowl just because your local cable system had an amplifier fail!). Finally, the average household watches television about five hours per day, so the exposure to outages is much higher than for voice telephone.

Taking these factors together, it is surprising that cable customers are willing to accept an availability as low as 99.7%. Clearly, this will not represent acceptable performance for the provision of switched voice, data circuits or PCS base station interconnect.

## THE SHORT HISTORY OF CABLE AVAILABILITY MODELLING

CableLabs' Outage Reduction Task Force

CableLabs' member companies wishing to enter not only the telephone business but the switched video market were concerned about the lack of perceived reliability of cable systems. They needed to quantify current network performance and develop tools for modeling the availability of systems in a systematic way so that different architectures could be compared quantitatively.

Given the historic lack of communication links among regional cable systems, the task force limited its studies to headends and local distribution networks. A key decision was to study only outages affecting two or more subscribers, and thus to eliminate the effects of individual drop cables and converters. Given that about half the individual subscriber outages in a typical cable system arise from drop and converter problems,[5] Cablelabs' results do not accurately reflect customers' perceptions of availability. On the other hand, the analysis tools developed are very valuable in comparative analysis of various distribution architectures.

CableLabs Availability Modeling

CableLabs' method was to gather data from participating companies on actual outages and their causes. From this data, they calculated average failure rates of various components involved. These failures rates were applied to a reference coaxial system architecture and classical reliability analysis techniques used to predict performance as a function of such parameters as the number of amplifiers and the reliability of various component types.

As a check on the model's applicability, the results were compared with actual recorded failure rates for systems with similar characteristics and the results compared with what the participants felt was adequate accuracy.

*Outage Reduction* was distributed to Cablelabs' member companies and highlighted at the Cable Tec Expo, among other technical gatherings. The computer model, in a generalized form, was distributed as a spreadsheet along with the study so that systems could apply it to their own situations.

Despite the tremendous effort put in by members of the Outage Reduction Task Force, the accuracy of the results are limited by the original data. In most, if not all, cable systems, outages are manually logged and accurate failure analysis and documentation is second in importance to restoring service. Thus, outage durations, the number of affected customers and the cause analysis are all of limited accuracy.

A final factor that must be mentioned is that the modelling and field data were taken on all-coaxial systems, so that the effect of fiber-optics on overall reliability was not determined.

A key finding of the group was that system powering problems dominated all other outage causes in most systems unless adequate standby powering was employed. Given that, the task force extensively analyzed utility power systems and techniques for minimizing utility outages and damaging transients. Many cable systems who are less than rigorous in deploying or maintaining standby power supplies might want to study this part of the document.

Rogers Cable's HFC Extensions

Nick Hamilton-Piercy and Robb Balsdon of Rogers Cable, in July, 1994,[6] used CableLabs' techniques to evaluate the theoretical availability of several HFC architectures. They found that various configurations had theoretical availabilities ranging from 0.9994 to 0.99985.

## REFERENCE HFC MODEL

Architecture

For purposes of calculating the affects of various reliability-enhancing techniques, the reference system architecture shown in Figure 1 was be used. This is typical of many modern cable systems where a central headend is connected by fiber to independent nodes, each containing small, non-interconnected coaxial distribution systems.

This is a scaleable model whose variables may include the number of cascaded amplifiers and other components, differing numbers of homes passed and different fiber interconnection options with the headend.

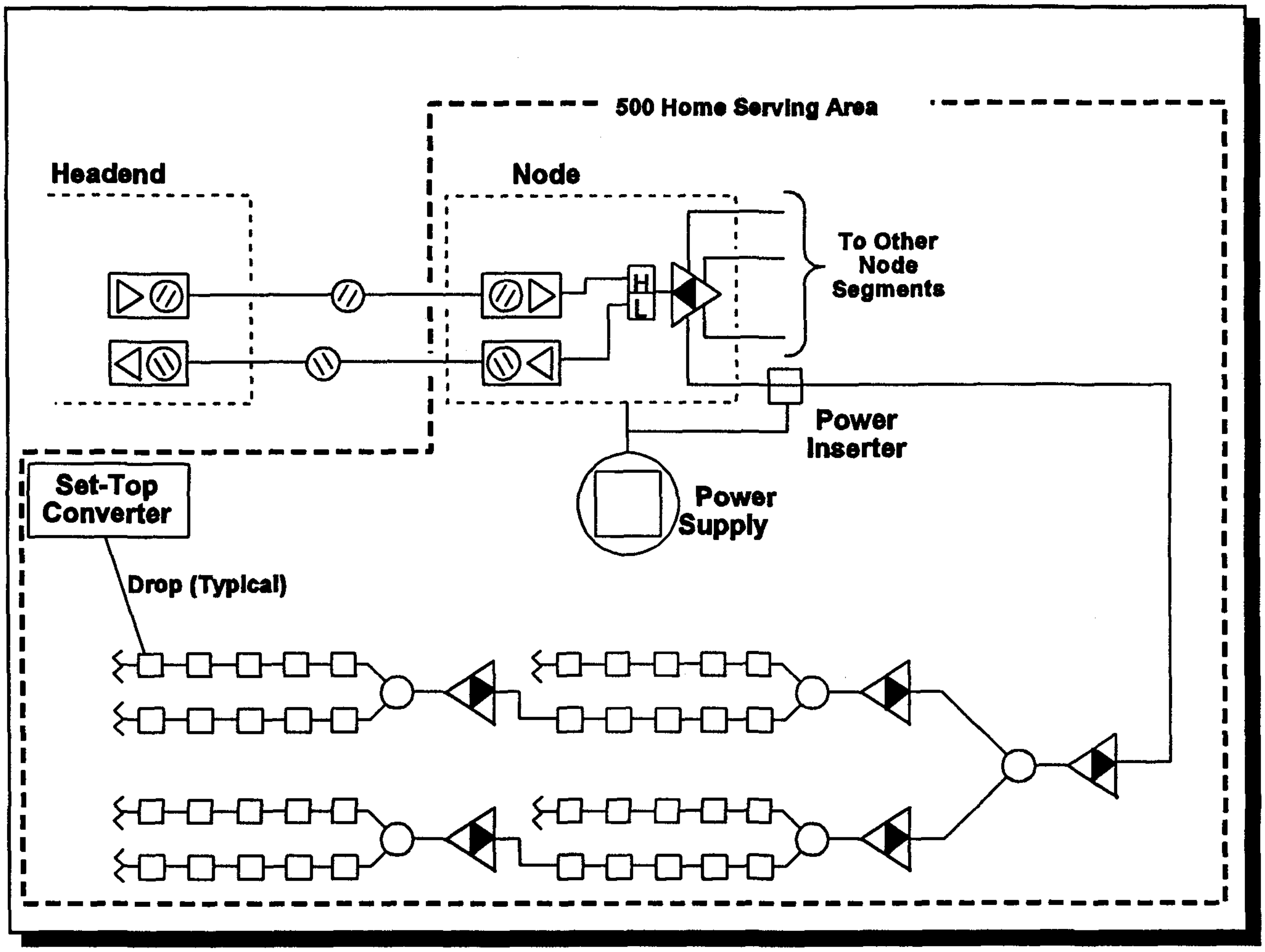


**Figure 1: Reference Architecture for Availability Analysis**

For modeling purposes, the following parameters were assumed:

- 500 homes per node.
- Five strand-miles of fiber between HE and node.
- One power supply per node serving area.
- Four coaxial trunks from each node.
- 160 four-port subscriber taps per node (78% tap efficiency).
- Three amplifier cascade past the node.
- 100 homes per mile density.

Repair Time Assumptions

Unlike the CableLabs and Rogers models, availability calculations included headend, plant, drop and converter failures and, therefore, true subscriber-perceived network availability. The algorithm also allowed for different average repair times ("mean time to repair" or MTTR) in the headend *vs* plant, which more closely resembles typical field situations.

For the reference model, it was assumed that the MTTR for plant failures was four hours (based on CableLabs data). It should be noted that this is much less than the seven hour MTTR found in an NRC study of telephone system fiber cuts, however the telephone cables cut were much larger, on average, than typical CATV cables, so the difference was expected.

One hour MTTR was assumed for headends, based on their proximity and accessibility to repair personnel. Actual headend MTTR will vary widely, of course. Large urban system headends may be manned part or all of the day, while small rural headends may be unmanned and remote from on-call personnel. This is one of the factors that is leading to consolidation of small headends into larger regional centers connected by fiber to large distribution areas.

## Component Failure Rate Assumptions

### *Headend*

An important component of headend reliability is its effect on viewers, *e.g.* if a subscriber is not watching a channel, he is unaffected by its failure. For modeling purposed, it was assumed that the average viewer would be affected by ten channels in one viewing session. In this respect, the model used was more liberal than that used by CableLabs, which counted any single channel failure as an outage.

In the author's model, headend failure rate was predicted by counting the number of pieces of equipment required to process ten channels, then multiplying that by average equipment failure rates. The average yearly failures rate for equipment was assumed to be 5%, the same rate used by CableLabs in their analysis.

The headend equipment configuration is not shown in figure 1, however, it was estimated that generating ten channels would require three satellite antennas, ten microwave receivers, five satellite descramblers and ten RF modulators. The mathematical model also includes three series-connected headend amplifiers (required to provide sufficient isolation between node-specific signals fed to individual F/O transmitters), each with a 3% annual failure rate. Although CableLabs assumed a 7% failure rate for trunk amplifiers and 5% for line extenders, it was felt that the lower rate was reasonable for an indoor mounted unit.

This combination results in a failure rate per year of 149%. With a one hour MTTR, the resultant availability of the headend is 0.99983. This failure rate correlates reasonably well with actual recorded outage data from a recently built headend where the average failure rate among 189 pieces of equipment was 6.3%.year.

### *Distribution Plant*

The distribution plant includes everything from the input of the headend optical transmitter to the subscriber tap port, but no drop components. The yearly component failure rates assumed for the initial analysis are as follows:

| Component | Yearly Failure Rate |
|---|---|
| F/O Transmitter | 7% |
| F/O Cable | 0.06%/mile |
| F/O Node | 7% |
| Power Supply | 5% |
| Passives | 1% |
| Taps | 0.5% |
| Connectors | 0.5% |
| Amplifiers | 5% |
| Coax Cable | 0.06%/mile |

These numbers generally follow the CableLabs recommendations with some exceptions:

- F/O transmitters and nodes were each assumed to have a reliability, based on complexity and heat generated, similar to trunk amplifiers.
- Rogers' data was extrapolated to predict F/O cable failure rate. Then, on the assumption that accidental damage represents the highest failure exposure

and similarly affects all types of cable, the same rate was applied to coax lines.

- Based on the author's experience, CableLabs' predicted 2% failure rate for standby power supplies, including batteries, was increased to 5%.

- The failure rate for trunk passives was taken from CableLabs' indication that this is a typical manufacturer's specification, despite their recommendation to reduce to 1/10 of that value.

- Tap failure rates were estimated at half the failure rate of trunk passives based on the lower average currents carried.

- Connector failures were only estimated in aggregate by CableLabs, with no scaling for quantity. The author's model uses an estimate of 0.5%/connector/year based on field experience.

- Since HFC systems typically use simplified amplifiers with fewer components than trunk stations, CableLabs' recommended failure rate for line extenders was applied to all post-node amplifiers.

It was assumed that the power supply is a typical cable television standby unit with approximately 2 hours of batteries capacity and no monitoring to alert operators that commercial power has failed. Given that, the model assumes a 30% probability of a power outage that will exceed the standby supply by 4 hours each year. Since there is a wide variance in the quality of local power grids, this parameter will, in practice, vary widely from location to location.

The estimated failure rate of the distribution system shown in Figure 1, using these component failure rates is 89.8%/year, as measured at the tap port feeding the most distant customer from the node. With a four hour MTTR, this results in an availability of 0.99959.

*Drop*

Although the details of the drop wiring are not shown in the figure, each drop is assumed to include the series connection of cable, four drop connectors and a ground block or splitter. Since CableLabs did not analyze single customer outages, their model does not suggest failure rates for drop components. Based on the author's experience, the following failure rates were assumed:

| Component | Yearly Failure Rate |
|---|---|
| Drop Cable | 3%/drop |
| F-Connectors | 5% |
| Splitter | 5% |
| Ground Block | 5% |

Given the above failure rates, a drop will experience annual composite failure rates of 28%, which translates into an availability of 0.99987.

*Converter*

Set top converters have historically been a frequent cause of single subscriber service outages. Although design and construction has improved in recent years, the newest converters are also much more complex. For modeling purposes, an 11% annual failure rate was assumed. Thus the converter alone will have an annual availability of 0.99995.

## REFERENCE MODEL AVAILABILITY CALCULATION

Using standard reliability/availability calculations, the series connection of headend-plant-drop-converter was predicted have an

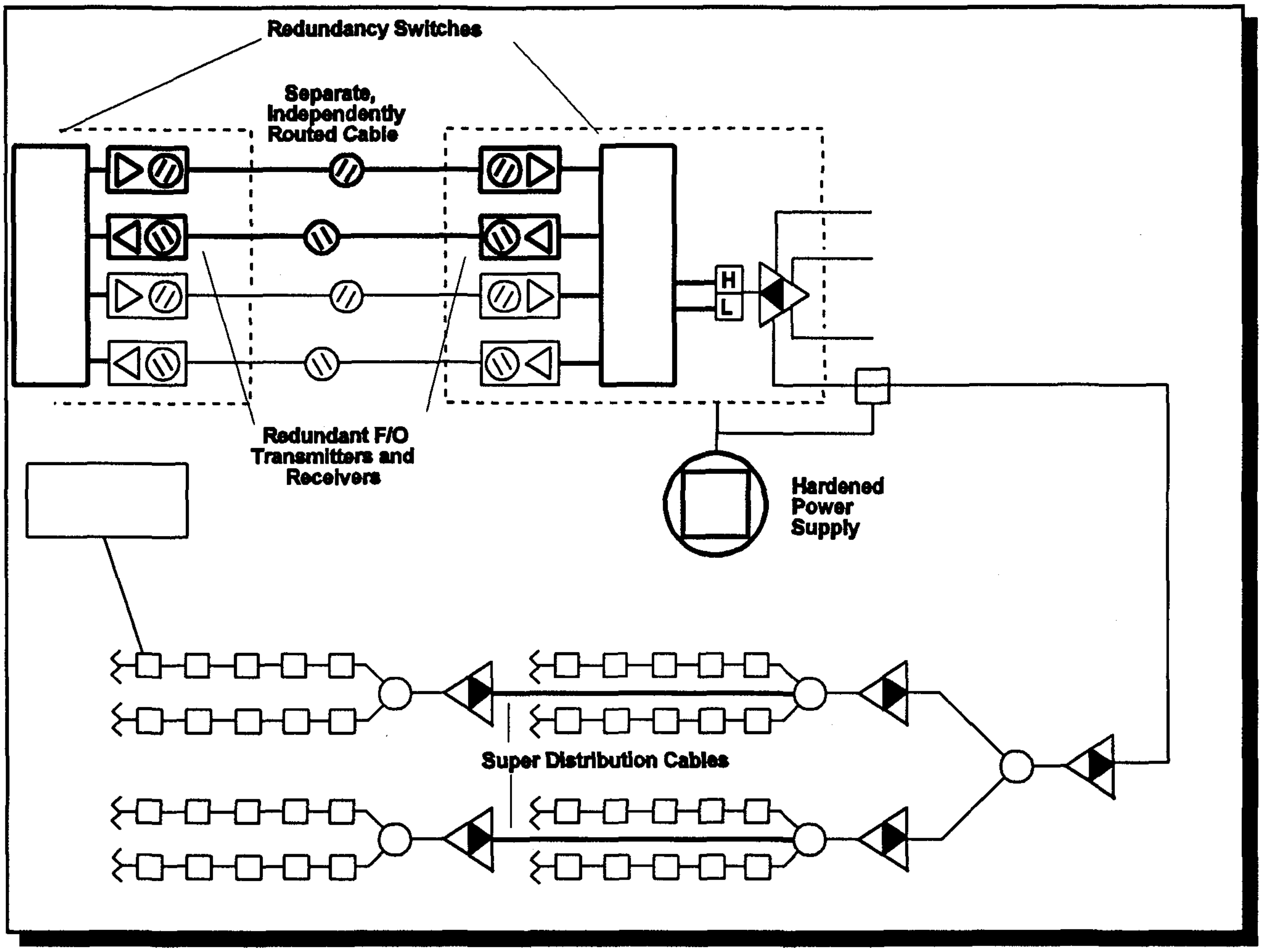


**Figure 2: Improvements in Reference Architecture**

absolute availability of 0.99924 equivalent to 398 minutes per year of outage.

A typical television subscriber, however, will only experience those outages which occur while viewing is occurring and, assuming a random distribution of failures, will perceive an availability of 0.99975. This represents only one twelfth of the outage rate that CableLabs found to be the critical viewer "irritation threshold".

It can be argued that, unless subscriber expectations change radically, this is sufficient reliability for video entertainment services. It may well exceed the reliability of today's telephone system as well, when plant and terminal equipment failures are added to known switch reliability. Nevertheless, it is worthwhile understanding what improvements could be made, if required.

## AN IMPORTANT CAVEAT

The above analysis theoretically applies to communications in either direction between subscriber and headend, as the component cascade is the same. In fact, however, communication in the reverse direction will be degraded due to two factors:

- Ingress signals which may cause communications failure, even if all components are properly working.
- Failures of terminal equipment or upstream transmitters which cause continuous transmission and thus block

upstream communications from other locations.

Analysis of these "soft" failures, as opposed to component failures, is beyond the scope of this paper, but must be considered when predicting overall system communications reliability.

## IMPROVEMENTS IN THE REFERENCE MODEL

Several modifications were analyzed to ascertain the degree of resultant improvement in the reference model's reliability. Those affecting the distribution plant are shown in Figure 2.

### Hardened Node Power Supply

Among the network sections, the lowest availability occurs in the distribution plant, whose predicted failures represent over half the total. About 1/3 of those are the result of commercial power outages in excess of the standby capacity of the node power supply. These may be eliminated or reduced by increasing supply capacity to greater than the duration of most outages, by status monitoring so that crews can be dispatched before the batteries fail, or both.

The regional Bell operating companies are generally opting for either eight hour battery capacity or inclusion of natural gas powered generators in each supply and, in either case, are deploying status monitoring.

### Use of Super-Distribution Architecture

Super-distribution[7] (also known as "express feeders") can be used to reduce the number of series-connected taps and connectors.

### Redundant Fiber Feeder Cable

A final increment of plant improvement can be gained by paralleling the fiber-optic transmitters, cable and receivers with an automatic changeover switch at the node. Given that cable reliability is primarily a function of accidental dig-ups and car-pole accidents, it is essential that the redundant cable not have any common routing with the primary cable.

These three improvements, taken together, improve the predicted plant availability from 0.99959 to 0.99983, equivalent to 88 minutes per year of outage. The overall system availability improves from 0.99924 to 0.99948.

### Reduced Headend MTTR

After the distribution plant, the next highest contributor to unavailability is the headend. Although its effect on individual customers is comparable to the plant, outages affect the entire customer base and so contribute to a disproportionate number of customer-hours of outage.

Possible tools for reducing down time include hot-standby redundancy with automatic changeover and/or 24 hour manning of the headend.

Even if the basic equipment reliability cannot be improved, reducing the MTTR from 1 hour to 15 minutes improves the headend availability from 0.99983 to 0.99996 which is comparable to many estimates of telephone switch performance.

### Improved Drop Connectors

From a reliability standpoint, F-connectors are inherently poor. Since there is no wiping outer conductor contact, good electrical performance, as well as mechanical strength, depends on having an appropriately tightened locking nut. Technician skills, temperature cycling and corrosion combine to create a low-reliability situation.

Although improved connectors have been available for a number of years, the basic failure mechanism remains. It is exacerbated by use of the connection as the primary means of turning service to homes on and off. When unmated connectors are left exposed to the weather, the resultant corrosion makes it even harder to assure a reliable connection.

There are several alternatives available and recently proposed. For instance, the German DeutcheBundesPost (DBP) uses a permanent, connector-less attachment of the drop at the tap and does the electrical connect/disconnect function in a side-of-building weatherproof housing[8]. An alternate approach has been proposed by a U.S. connector manufacturer which has suggested building a sealed switch into taps, so that the drop-to-tap connection can be made permanent[9].

Reducing the connector failure rate from 5% to 2% per year would increase the availability of individual drops from 0.99987 to 0.99993.

Improved In-Home Descrambling Hardware

With the planned increase in complexity in set-top converters necessary to support on-screen graphics, digital decompression, etc., there is little reason to believe that the reliability of these components will increase significantly without a disproportionate increase in cost.

If the FCC, as part of its rulemaking implementing the consumer interface section of the Cable Act of 1992, mandates the availability of set-back decoders, they should theoretically be more reliable because tuners will not be required. While such improvement was not considered in this availability estimation, system reliability may benefit from it in the future.[10]

Achievable Availability

Taken together, the above measure allow overall absolute availability to be increased to 0.99967, equivalent to 175 minutes per year of outage. The perceived availability to a typical video subscriber is 0.99989 or 58 minutes per year, roughly one thirtieth of the outage time found by CableLabs to be the threshold of

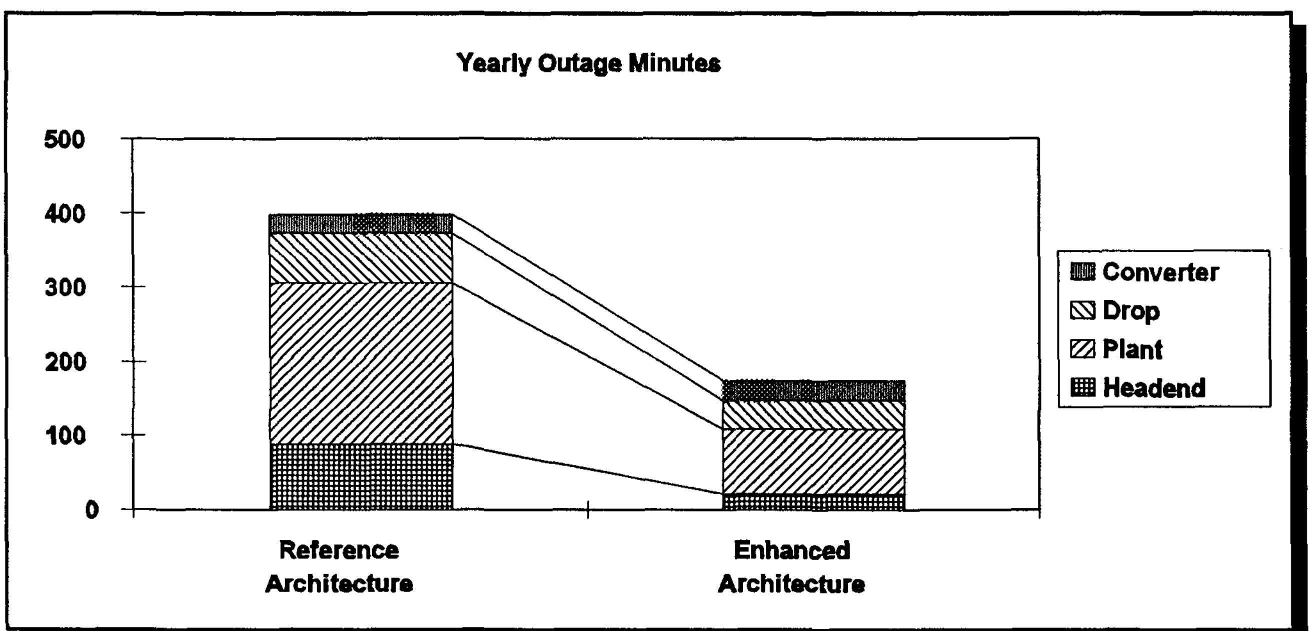


**Figure 3: Predicted Outage Improvement with Enhancements**

serious subscriber dissatisfaction.

Figure 3 shows graphically the predicted minutes of outage for the reference model and that attainable with the improvements discussed.

## WHAT'S WRONG HERE

The problem with this analysis is that it is based on failure rate data that is of marginal accuracy, at best. Until comprehensive failure information is gathered or use of automatic monitoring becomes common, field data will be suspect. Under typical field conditions today, many small outages go unreported and/or not analyzed. Individual customer outages are not included in failure analysis by most systems.

Compounding this, cable operators have not generally required MTBF data from their suppliers, so that, in many cases, not even manufacturers' predicted data is available.

As a result, while the analysis tools developed by CableLabs are invaluable for evaluating relative performance of various architectures, meaningful predictions of actual network performance will only be possible when much-improved component data is available.

## CONCLUSIONS

Based on the estimated component failure rates listed earlier, this analysis predicts that typical small-serving-area hybrid fiber/coax networks, coupled with reasonable repair times, can achieve outage times sufficiently low to satisfy customer expectations for current video services.

Various measures, particularly hardened power supplies and improved headend availability, can be undertaken to considerably increase availability. Using all of the measures described, a perceived availability of 0.9999 is predicted.

## ADDENDA

Reliability/Availability Terms

*Failure Rate*: The percentage of devices which fail in a specific period of time (specified as %/year in this paper) and expressed with the symbol $\lambda$.

*Reliability*: The probability of failure in a specific time period, mathematically:

$$R(t)=e^{-\lambda t}$$

For a series connected system, the net reliability is the product of the reliability of the individual components:

$$R_S(t)=R_1(t)\ R_2(t)\ .\ .\ .\ R_n(t)$$

*MTBF*: Mean time between failures, mathematically

$$MTBF=\frac{1}{\lambda}$$

*MTTR*: The mean time required to restore operation after a failure.

*Availability*: The decimal amount of time that a mechanism (in this case a cable system) is available to the user. Mathematically:

$$A=\frac{1}{1+\frac{MTTR}{MTBF}}$$

*Outage Time*: Time the system is unavailable in a specified time period. For instance:

$$Outage\ (\mathrm{min}/yr)=minutes\ /year\ (1-A)=525{,}600\ (1-A)$$

References

1. The FCC established the NRC in 1992 in response to several large failures of the public

switched telephone network (starting in 1988) which affected thousands of telephone circuits. Its scope was limited, however, primarily to failures of switches and cuts of major inter-office fiber cables. The re-constituted council (with cable television representation) is just beginning to deal with the issues raised by new local distribution means.

2. See the Glossary for an explanation of the relationship between reliability, repair time and availability.

3. To, Michael and Neusy, Philippe, "Unavailability Analysis of Long-Haul Networks", *IEEE Journal on Selected Areas in Communications*, vol 12, no 1, January, 1994.

4. *Outage Reduction*, an internal publication of Cablelabs, Boulder Colorado, distributed to all CableLabs member companies (who together serve 85-90% of the nation's cable subscribers). The first chapters of this manual and study were distributed in September, 1992.

5. Based on the author's experience and analysis of trouble call resolution data from a large multiple system operator.

6. "Network Availability and Reliability", *Communications Technology*, July 1994.

7. Super Distribution has the added advantages of allowing increased physical spacing between amplifiers and of decreasing the effects of multiple signal reflections among the imperfectly matched coaxial components.

8. *Breitbandverteilnetze de DeutschenBundespost, 2.,überarbeitete und erweiterte Auflage*, Dipl.-Ing Hans Steckle, published by R.v.Decker, Heidelberg, 1988 (German). Like the U.S. telephone network, the DBP cable system terminates at the building entrance, where all provisioning takes place. Cables between the dwelling and tap are generally of semi-rigid construction with solid copper jacket and center conductor and are hermetically sealed at the connection point with the tap.

9. "Approaching the 'Seamless Drop'", Brian Bauer, Raychem Corporation, *Communications Technology*, June, 1994.

10. *First Report and Order In the Matter of Implementation of Section 17 of the Cable Television Consumer Protection and Competition Act of 1992*, Federal Communications Commission, released May 4, 1994, paragraph 34.

## Using Cable TV's Hybrid Fiber Coax (HFC) Infrastructure for Data Communications

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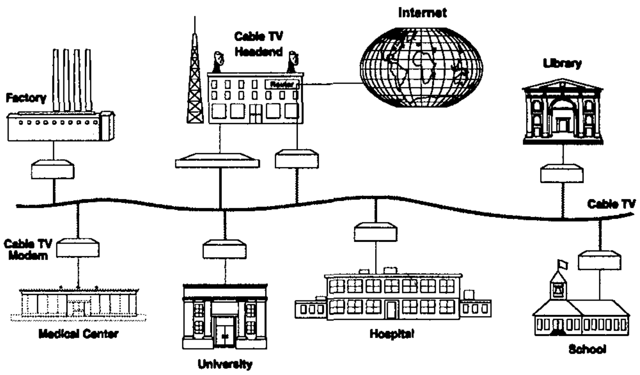
### ABSTRACT

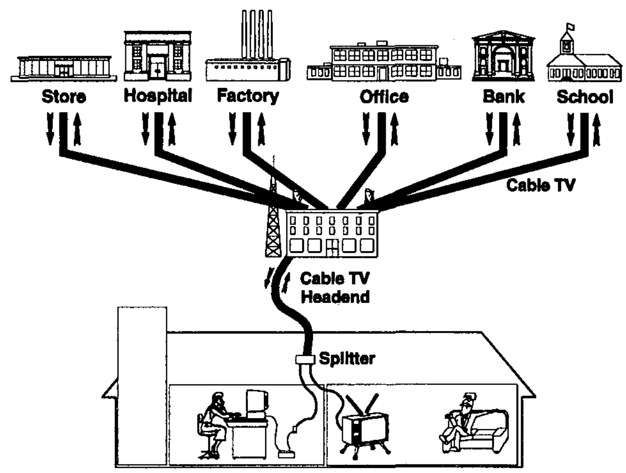
*Using the available cable TV infrastructure, cable TV modems can interconnect homes, offices and municipal governments within our towns and cities. However, complex technical issues must first be overcome to create city-wide data over cable TV services offering cable operators a viable new source of revenues. This paper provides a conceptual overview of the cable TV operating environment, the types of information services that can be offered over cable TV, and a description of the enabling technology—cable TV modems—which make high-performance data communications over cable TV a reality.*




**Typical Cable TV High-Speed Data Network**

### INTRODUCTION

Today's cable TV infrastructure connects businesses, homes, shops, schools and government organizations city-wide. Cable TV internetworking is simple, because network connections are distributed, unlike telephone networks which only support point-to-point links requiring complex modem networks. The types of information services that are being provided over cable TV are as varied as the homes and businesses the cable passes. Internet access services, local area network interconnectivity, work-at-home, multimedia database services and multimedia conferencing are just a few of the many applications communities are employing to run their companies, teach their students, occupy their leisure time and improve productivity. Generating significant new revenue streams from the existing cable TV infrastructure, cable operators are first provided the enabling technology—cable TV modems—enabling the necessary high-performance yet economical city-wide data connectivity.




**Work At Home Concept**

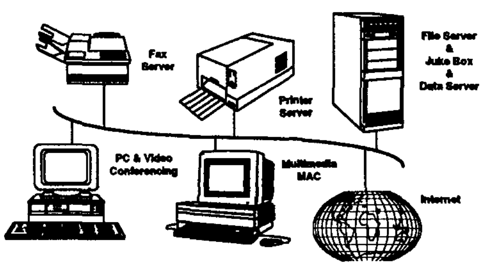
### OPPORTUNITY OF DATA COMMUNICATION SERVICES OVER CABLE TV

Providing information services over metropolitan cable TV plants places unique operational requirements on the communications equipment used. These requirements are based on the two fundamental areas of data communications: the physical layer (PHY) and the media-access layer (MAC).

The operating characteristics of the transmission medium are defined by the physical plant, and in the cable TV world this includes the cabling, splitters, diplexers and amplifiers. The media-access layer defines the control of access to the physical layer that is shared among many connected users. The cable infrastructure is powerfully attractive for providing information services. Initial technical certification of the environment must first be addressed to ensure technical and commercial success for the cable TV operator.

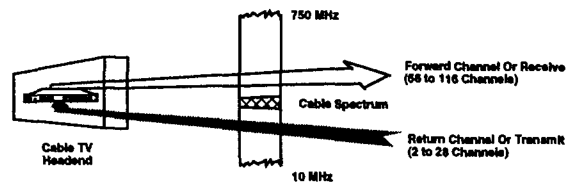
Technology which has been designed from the ground up to operate within the requirements imposed by the physical and media-access control layers provides information services over cable TV reliably. This technology is now available to provide the required high-performance levels over diverse operating conditions—and at market-driven prices.

The evolving Client/Server computer network model and Internet utilization patterns confirms that the source of information for data, video conferencing, application services and the Internet is distributed throughout the network. Therefore, it is clear not to assume that all information can originate at the headend and propagate down the forward channel.




**Client / Server Computer Model**

Data rates on both the forward and reverse channels must provide high transfers to support the distributed computer and multimedia paradigm. Ten megabits per second (10 Mbps) and above on the forward and reverse channels provide the necessary data rates to run the multimedia applications of the 1990s.

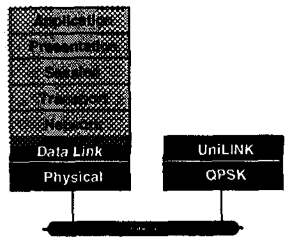



**Cable TV Headend 2-Way Infrastructure**

The many types of information services a cable operator provides for incremental revenue streams—video conferencing, multimedia databases, work-at-home, etc.—each have their own requirements for data rates, latency times and periodicity of access. Because of the scarcity of return channels to the headend, these different levels of services must all be provided within the same 10 Mbps streams and 6 MHz channel.

The diverse information services requirements are as follows: Efficient resolution of access contention and recovery from congestion is provided, because the coax cable which connects multiple users to the headend is a shared medium. Fair and stable access under varying traffic conditions is also required. Additionally, the technology itself is very easy to install, configure and maintain, so that the cable operator controls access, operational parameters and configures the network remotely. Operational visibility allows for easy troubleshooting and fault isolation. Security mechanisms prevent illegal tampering and use. Finally, a cable TV media access control protocol—which acts like a traffic cop directing traffic on and off the network—allows connected nodes to exchange multimedia information at very high performance levels.

A MAC-layer protocol for the metropolitan-area environment which addresses the above requirements and is being successfully installed on many cable TV systems worldwide is named UniLINK™ and was developed by LANcity Corporation.




**OSI Model As Related To UniLINK™ Model**

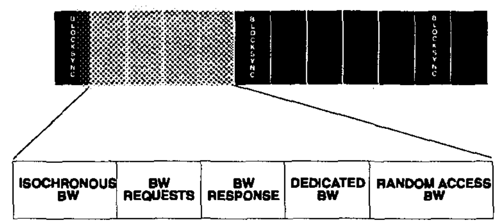
Offering enhanced digital signal processing modem technology, this cable TV protocol is uniquely positioned to provide easily installable, high-performance information services to homes.

## CABLE MODEMS FOR THE METROPOLITAN ENVIRONMENT

The following sections provide an overview of the modem technology which allows a high degree of operating efficiency in the metropolitan cable TV environment.

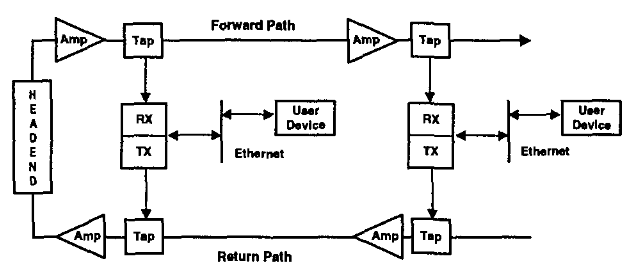
## UNILINK ADAPTIVE ACCESS PROTOCOL

UniLINK™ is a media access control protocol which addresses requirements for providing information services to the cable TV user over metropolitan areas. It provides two modes of operation, enabling the many different types of applications described earlier to operate efficiently and reacts to changes in data traffic loads by reallocating the available bandwidth to relieve congestion situations. Both shared contention and dedicated reservation-based access are allowed within the same operational channel. Contention access provides almost immediate access under light loads but tends to bog down with higher levels of data traffic because of collisions and backoffs. UniLINK™ has the unique ability to switch from contention- to reservation-based access to help relieve congestion during high traffic loads. The protocol supports a simultaneous mix of both types, and it switches dynamically as necessary to maintain peak efficiency on the network. A "traffic cop" allows traffic to flow at its own pace until congestion occurs, at which point it starts directing who has access to the roadway until the congestion clears. A similar concept exists with the UniLINK™ Access Method.




**UniLINK™ Access Method**

Both forward and reverse data rates are provided at 10 Mbps to offer standard local area network levels of service to users. Increased distances do not degrade the overall throughput, because each node determines and compensates for its distance from the headend—especially important because of the distances involved in cable TV systems. Each node times its transmissions on the network to timing reference packets provided by a single pacer node. Any node may serve as the pacer, which is determined by an automatic election process. Multiple packets can be outstanding on the network, precisely following each other, because of the synchronous and distance compensating nature of the protocol, which allows efficient channel use. The data communications channel timing system can be compared to an Interstate Highway System, which is efficient because it allows multiple vehicles to travel simultaneously.



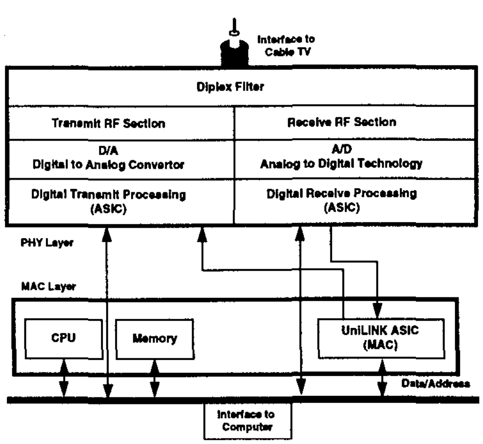

**Data Communication Over Existing Cable TV Infrastructure**

Both an Interstate Highway System and the information superhighway would be of limited usefulness if only a certain type of vehicle were allowed to travel on it. So the UniLINK™ protocol has been designed to service the traffic patterns that will be created by the varying applications on the cable TV network. Constant bit rate (CBR) with no jitter for voice applications, variable bit rate (VBR) with limited jitter for multimedia applications, and available bit rate (ABR) for basic data services can all be supported in the same 6 MHz channel.

The UniLINK™ protocol addresses the distance, changing data traffic, multiplicity of applications and congestion relief so prevalent in the cable TV networking environment.

## ADAPTIVE MODEM FOR THE CABLE TV ENVIRONMENT

The new generation of cable TV modems that has evolved allows robust, high-performance within a harsh RF cable operating environment. The field-proven UniLINK™ protocol provides technology that compensates for various aspects of signal degradation using adaptive equalization mechanisms. The advantages of using the UniLINK™ protocol are that it provides periodic timing references on the network that can be used to sample signals on the cable TV HFC cable at specific times, regardless of whether actual payload data can be derived from the signal.




**Data Cable TV Modem Architecture**

The process used to allow both the forward and reverse channels to be compensated and is proven to independently compensate for the effects of the forward and reverse paths through a cable TV plant with no equalization training overhead for individual packets.

Cable plant characteristics will vary over time. Each node can periodically recheck its forward- and reverse-channel tilt and delay coefficients to provide peak performance levels at all times. Equalization can be performed every few hundred milliseconds to take into account "channel surfing" effects.

Cable TV modem technology described herein has been implemented in two Application-Specific ICs (ASICs) totaling 200,000 gates, allowing for compensating for the following channel characteristics:

| Operating Characteristic | Forward Channel | Reverse Channel |
|---|---|---|
| In Channel Group Delay Error | <= 240 nsec | <= 800 nsec |
| In Channel Tilt | <= 6 dB | <= 6 dB |

**Cable Plant Adjustment Range**

These channel characteristics can and most likely will be different for each drop on the network in both the forward and reverse paths, because a signal must traverse a unique path from the drop to the headend and conversely from the headend to each individual drop. Fixed, predetermined and static equalization information is therefore precluded because of the individual path variations that can change over time.

## TODAY'S CABLE TV MODEM REQUIREMENTS

The cable TV modem technology incorporates a 10 Mbps QPSK transmitter and receiver and is designed to operate on a subsplit cable TV plant, allowing operation in the 5 to 42 MHz frequency range. Because of the mentioned operating characteristics introduced by the forward and reverse channel paths, the cable TV modem incorporates equalization filters in the transmitter and receiver to provide for dynamic equalization to compensate for these characteristics. These filters allow the cable TV modem to function with severe channel tilt, reflection, and group-delay effects.

The cable TV modem uses fully independent transmit and receive equalization filters, so that individual path errors to and from the headend can be fully compensated. Additionally, a transmit power adjustment on each transmitter and a received power sense circuit allows each modem to sense and adjust its transmit power in reference to a node at the headend, so that dynamic power adjustment can take place network-wide. A wide dynamic range on both transmit power and receive range allows compensation for widely varying levels in both the forward and reverse signal paths.

Much of advanced modem functionality has moved from the analog domain into the digital domain, which reduces manufacturing complexity and manual tuning requirements, and allows powerful digital signal processing techniques to be used to compensate for operating conditions. The modem transmitter consists of a high-speed Digital-to-Analog converter, a lowpass filter, a PIN diode attenuator and an output amplifier. The transmit baseband filter, transmit frequency generation and QPSK modulation functionality are implemented inside of a digital ASIC.

The Digital-to-Analog converter (D/A) has been selected to provide adequate signal-to-noise ratio over the entire transmit band and operate at a high enough sample rate to make the reconstruction filters achievable.

The input to the transmit equalization filter is the I/Q data at a 5 megasymbol-per-second rate. The filter coefficients are derived from the equalization process. The carrier is generated using a numerically controlled oscillator using a sine and cosine generator. The sine and cosine outputs of the numerically controlled oscillator are sent to the modulator where they are mixed with the I & Q channel data. The output of the modulator is sent to the digital to analog converter in the modem transmitter analog section.

The modem receiver also relies on digital technology for the implementation of major functional blocks, including the Automatic Gain Control (AGC) circuit, the receiver equalization filter, symbol clock recovery and the QPSK demodulator.

## PLUG & PLAY FUNCTIONALITY

Cable TV data modems must adhere to the philosophy of simple installation and maintenance, coined "Plug in and walk away." The only requirements of the end user site is to make the physical connections to electric power and the cable TV. The node then begins frequency surfing to find its operational channel.

Other installation considerations must be addressed by the cable TV operator to allow plug-and-play functionality. These issues are minor and are associated with the cost of doing business, similar to installing a headend signal processor or frequency modulator to provide additional or upgraded video services. The only end user requirement is to make the physical connections to the power, cable TV and computer.

## COST EFFECTIVE DESIGN

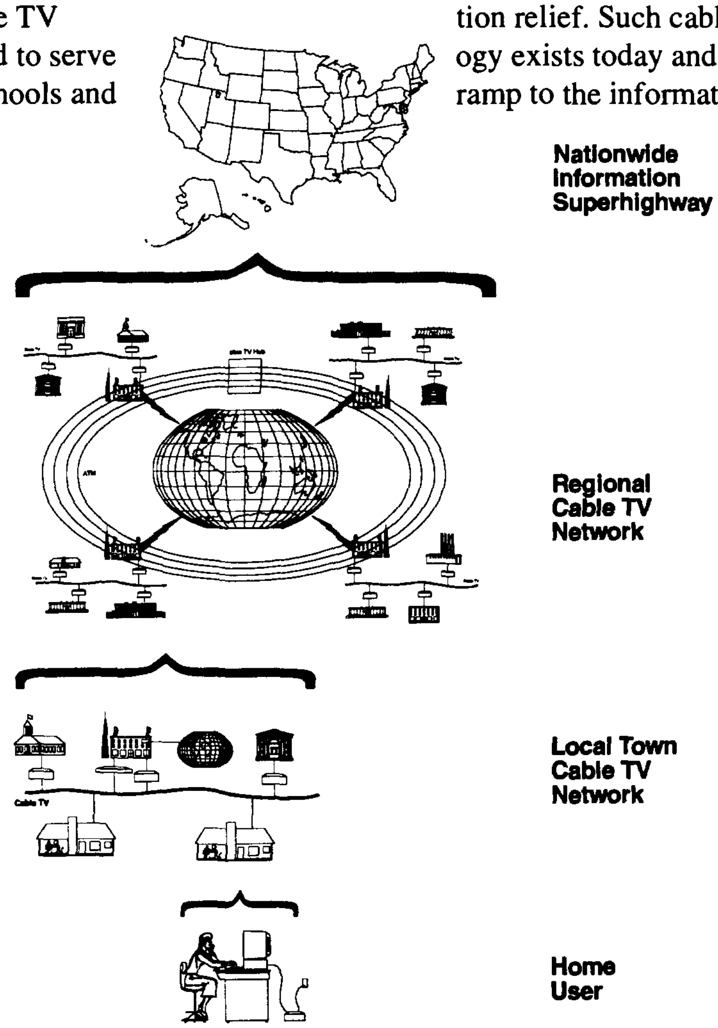
Product technology has been designed to meet the demands of the consumer or "at-home" market. A good design reduces the part count by increasing the level of integration through ASIC technology and utilizes components which have application in the consumer electronics market place. At the heart of the design is the Digital Signal Processing ASIC, which provides a high level of channel equalization and Automatic Gain Control functionality. The RF section uses components widely available for the cable TV set top box and cellular telephone market.

## LIFETIME COSTS

The cost of equipment is a small percentage of the lifetime cost to maintain and troubleshoot a network that covers metropolitan distances. Easy, low cost installation coupled with remote operational information easily accessible through network management are the keys to prevent high, long-term maintenance costs not apparent at the time of purchase.

## SUMMARY

The existing cable TV infrastructure is poised to serve homes, businesses, schools and governments with a wide variety of information services, generating a significant, new source of revenues for today's cable TV operators. To ensure commercial success, the requirements imposed by the two fundamental areas of data communications—the physical and media-access layers—must be designed to provide high performance levels over wide operating conditions. Key operating characteristics include high data transfer rates on both the forward and reverse channels at or above 10 Mbps to accommodate two-way data communications; a 6 MHz channel to compensate for the scarcity of return channels and to support the unique requirements for data rates, latency times and periodicity of access; and a digital signal processing protocol to address the prevalent issues of distance, changing data traffic, multiplicity of applications and congestion relief. Such cable TV modem technology exists today and can serve as the on-ramp to the information superhighway.




**Business Model**

# Video Server Architecture

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Time Warner Cable

*Abstract*

*Video-on-demand and other interactive television services are driving the development of video server technology. By their very nature these video servers are multiprocessor systems. This is especially true when the server is required to provide hundreds, thousands, or even hundreds of thousands of video streams. This paper reviews a number of the proposed video server architectures. These architectures differ in some key areas and this paper compares their advantages and disadvantages. Issues regarding their impact on software architecture, reliability, and cost are also discussed. Finally, a summary of areas for future work is presented.*

## INTRODUCTION

Video-on-demand and other interactive services such as interactive games and interactive shopping are driving the development of video servers [1,2]. Video servers are the computer systems that provide the storage and playback of MPEG compressed video data for interactive services in a broadband network environment. As they are designed to serve a large number of subscribers, video servers are shared resources located at the headend of the cable distribution network [3,4,5].

Figure 1 is a simplified diagram of a hybrid fiber/coax based interactive cable television system. This diagram shows the relationship of the video server to the other components in the end-to-end cable system and identifies the distribution headend and distribution network components. The headend equipment includes one or more video servers and an ATM switch or switching network. For simplicity, the ATM switch is shown as a single switch, but actually may be a network of ATM switches. The number of ATM switches required depends on the service requirements. Alternatively, its function may be subsumed as part of the video server itself. The distribution network is made up of the fiber-optic links between the headend and the neighborhood area network nodes (NAN) and the coax links from the NAN to the subscribers' homes. The home communications terminals (HCT) are located in the subscribers' homes and decode the MPEG data sent to it from the video server at the headend. Not shown in Figure 1 are the modulators, demodulators, amplifiers, and combiner/splitter networks that are also part of the distribution network. The traffic flow in the distribution network is asymmetric as the bandwidth required from the headend to the HCT for MPEG data is much greater than the return path for subscriber requests.

The role of the video server is to provide storage and playback of the video data and to respond to subscriber requests such as "pause", "fast forward", or "view the next product". The ATM switch is responsible for routing data from the video servers to the appropriate neighborhood nodes. The MPEG compressed video data is transmitted using ATM transport [6]. The ATM data is modulated using QAM modulation and transmitted over fiber from the headend to the neighborhood node. At the neighborhood node it is converted from an optical signal to an electrical signal for distribution over coax to the subscriber homes. These signals are then received by the home communications terminal where the video data is decompressed and displayed on the subscriber's television.

The actual number of video servers required in Figure 1 will be determined by the capacity of an individual server and the number of active HCTs being served. In the larger video server architectures the switching function provided by the ATM switch may be subsumed by the interconnection network of the video server.

Video servers are multiprocessor systems. Uniprocessor systems are unable to sustain the high data rates necessary for this type of application. A variety of multiprocessor architectures have been proposed to meet the high bandwidth requirements of a video server. The following sections will compare and contrast these video server architectures.

## VIDEO SERVER ARCHITECTURES

All video servers are implemented by some form of multiprocessor architecture. Video server architectures are characterized by two key attributes the type of CPU interconnect that is used and the video data path from disk to distribution network. Based on these two characteristics video server architectures fall into one of three categories:

- Symmetric Multiprocessing (SMP)
- Massively Parallel Processor (MPP)
- Loosely Coupled Computer (LCC)

The following sections discuss these architectural categories in detail, their advantages and disadvantages, and implications for software architecture.

### Symmetric Multiprocessing (SMP) Architectures

Shared memory SMP architectures are characterized by a high-speed system bus interconnecting CPUs, memory, and disk and network I/O subsystems. Figure 2 shows a diagram of an SMP architecture. This figure shows only four CPUs, but more are possible. Because the memory and I/O devices are shared equally among all of the CPUs in the system these architectures are called symmetric multiprocessor systems. Typically, these systems scale from one CPU to as many as 36 CPUs. Examples of SMP architecture include the Silicon Graphics, Inc. Challenge server systems and the Sun Microsystems, Inc. SPARCserver systems.

In the SMP architecture, the processing load is uniformly distributed across

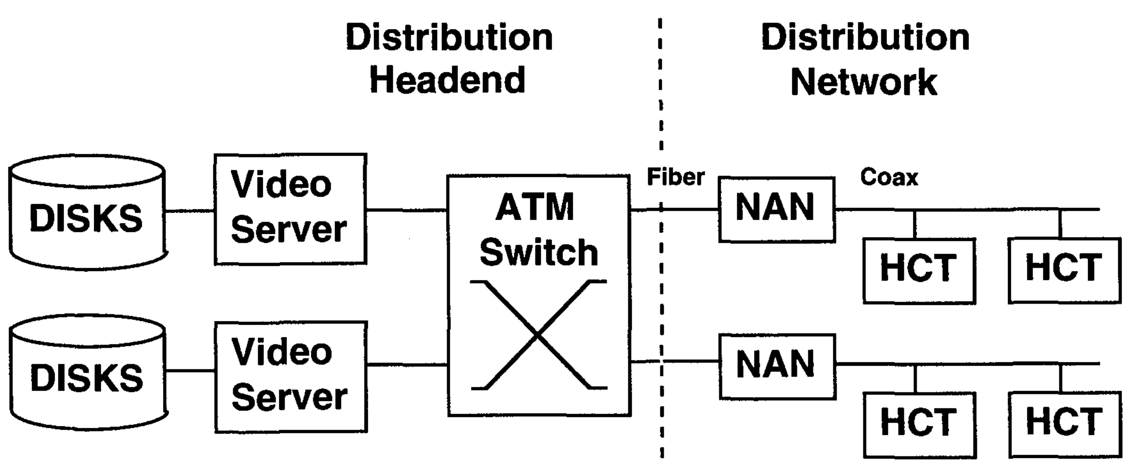


Figure 1 - Interactive Cable Television System

all of the CPUs in the system. Scheduling algorithms in the kernel allocate processing load to unoccupied or lightly loaded CPUs. All CPUs have equal access to the physical memory and see a uniform memory image. Data written to the shared memory by one CPU is immediately available to all the other processors. Software processes, both kernel and user, are able to run concurrently, and can be switched arbitrarily among any of the processors in the computer. Since all of the CPUs are operating out of shared memory, access to critical data and code is protected by locking mechanisms that prevent concurrent access.

Data flow for video data in an SMP system is straightforward. Video data is read from the disk into a buffer area in the shared memory of the system. The data is then transferred from this buffer area to the ATM network interface for transmission to the ATM switch. In this way, the video data crosses the system bus twice in the process of moving from storage on the hard disk to network transmission. As a result the system bus must have relatively high bandwidth to support the video traffic for all streams active in the system as well as all other general processing requirements. An example is the Silicon Graphics, Inc. Challenge server systems, which have a system bus bandwidth of 1.2 GBytes/second.

Massively Parallel Processor (MPP) Architectures

Massively Parallel Processor architectures, as the name implies, use large numbers of processing units. The processors are independent but are interconnected through high-speed networks. Unlike the SMP architecture, each processor in the MPP architecture has its own memory and I/O devices. This memory and I/O devices are local to each CPU and cannot be seen directly by other CPUs in the system.

In large scale MPP systems the issue of inter-processor communication becomes critical. To efficiently share information between processors it is important to have a high-speed interconnect between them. It is impractical to build a high-speed interconnection network that connects every processor to every other processor through a point-to-point network. Rather an approach that minimizes the latencies between processors and scales well is the hypercube architecture. Typically these systems scale from as few as 16 CPUs to as many as 8,192 CPUs. An example of an MPP architecture

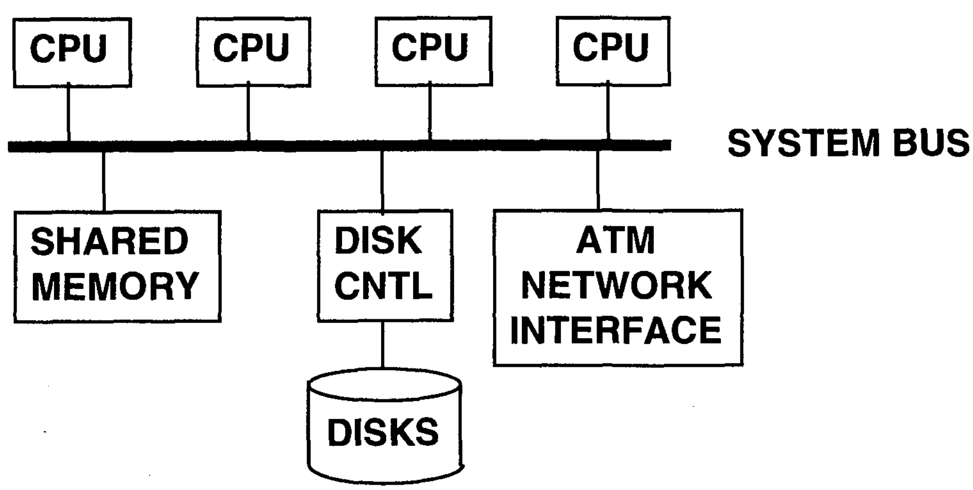


Figure 2 - Symmetric Multiprocessing Architecture

for video server applications is the nCube, Inc. MPP Media Server [7].

In the hypercube architecture each processor is connected to $N$ of its neighbor processing units, where $N$ is the order of the hypercube. The order $N$ determines the number of processors in the system, $Number\ of\ Processors = 2^N$. The order $N$ also determines the maximum number of links and the maximum delay between any two processors, *Maximum number of links* $= N$. Figure 3 shows the hypercube MPP architecture for orders one, two, and three. Each CPU has its own memory and I/O interfaces. The I/O interfaces will connect either to the disk controller, ATM network interface, or hypercube interconnect. The video data flows from the disk connect to a CPU and is routed either through the hypercube interconnect to the appropriate CPU or directly out through the ATM network interface. When the video data is routed through the hypercube interconnect, it may not be necessary to have an ATM switch in addition to the video server. Since the video data managed by an individual CPU is significantly less than for the video server as a whole, the bus bandwidth for an individual processing unit is correspondingly lower. However, the hypercube interconnect must be of relatively high bandwidth as it must carry a fairly high percentage of the video traffic. For example, the nCube system has a bandwidth of 2.5 MBytes/second for each link in the hypercube interconnect.

## Loosely Coupled Computer (LCC) Architectures

Similar to MPP architectures, LCC architectures are characterized by many independent processing modules containing CPU, memory, and I/O subsystem, linked together through a high-speed interconnect network. Unlike MPP architectures, the LCC architectures do not scale to as large a number of CPUs and use low-cost, off-the-shelf components such as Intel 486 or Pentium-based computers. LCC architectures scale to hundreds of CPUs rather than the thousands found in MPP architectures. LCC architectures typically use a mesh based interconnect rather than a hypercube. Examples of LCC architectures include the Intel Corporation Scalable MultiServer [8] and the Digital Equipment Corporation Interactive Information Server [9].

Figure 4 shows a diagram of a LCC

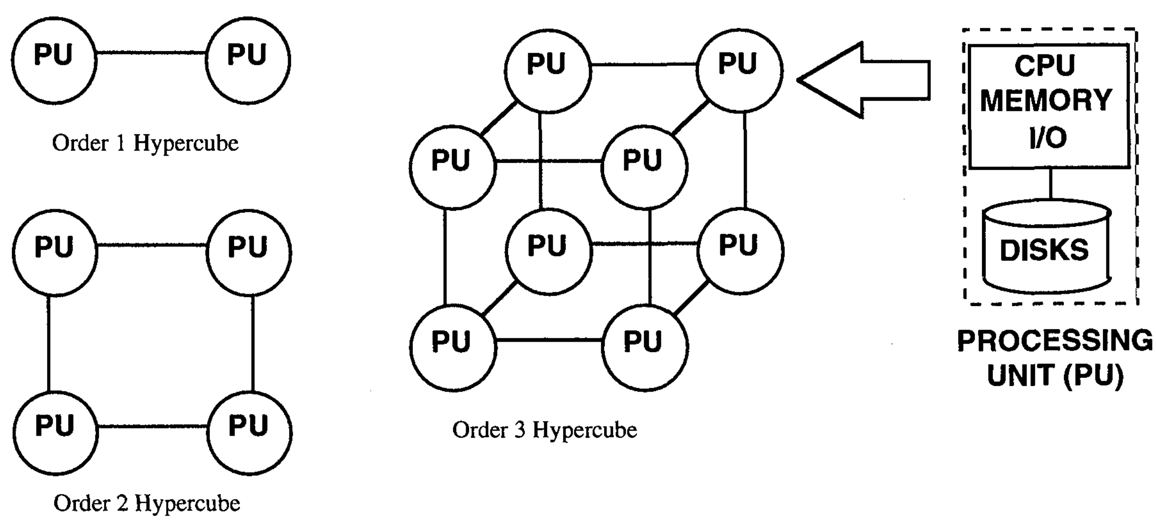


Figure 3 - Massively Parallel Processor Architecture

architecture. In this architecture, each of the processing units represents an Intel 486 or Pentium-based computer. These low-cost computers are connected through a two-dimensional grid, implemented as a cross bar switch or mesh network. This increases the network delay between processing units and decreases the size to which the system can be scaled. Large scale crossbar or mesh networks are impractical to build.

The data flow of video data in the LCC architecture is very similar to the data flow in the MPP architecture. Here, video data is transferred from the disk to the mesh interconnect for routing to the appropriate processing unit for distribution to the ATM network. In some instances, the video data for an individual movie or video clip will be spread across several of the processing units in the system for redundancy and improved concurrent access.

## ARCHITECTURE COMPARISONS

The three video server architectures, SMP, MPP, and LCC, also impact four key areas:

- software architecture
- scalability
- reliability
- cost

The following sections compare the three architectures' impact in each of these areas.

### Impact on Software Architecture

The primary difference between the SMP, MPP, and LCC architectures that affects the video server software architecture is the memory and CPU interconnect; the shared-memory model of the SMP architecture versus the local-memory model of the MPP and LCC architectures. In the SMP architecture there are multiple processors looking at a unified physical memory space. In other architectures each processor is only looking at its own local memory. This single aspect, more than any other drives the video server software architecture.

The high-speed system bus allows much faster interprocessor communication in the SMP architecture than in either the MPP or LCC architectures making data sharing

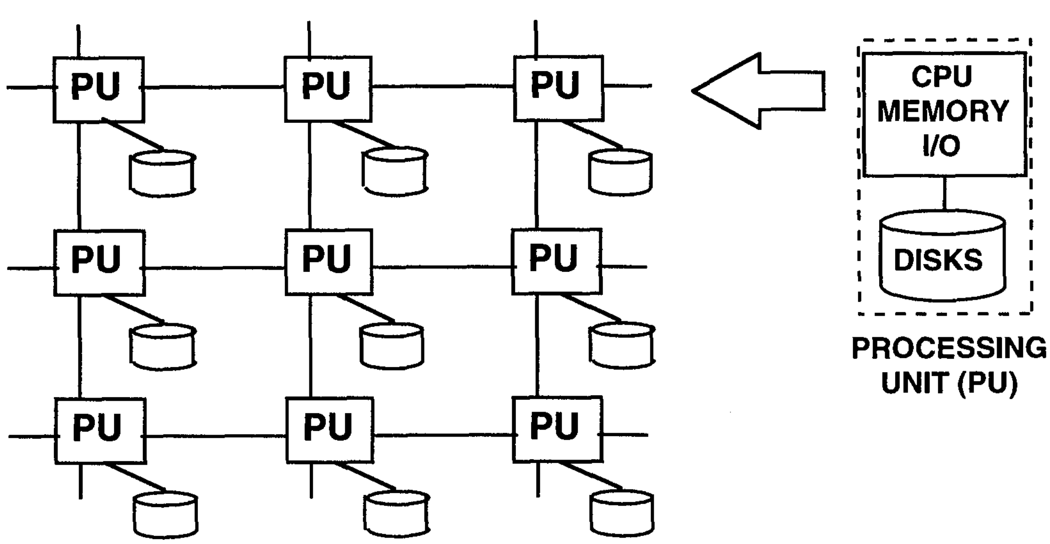


Figure 4 - Loosely Coupled Computer Architecture

between processors is much simpler and faster. The SMP architecture is easier to program and provides faster synchronization between processors and processes. The other architectures have a more difficult time providing a simple multiprocessor programming environment and fast process synchronization.

In the SMP architecture the application software is shielded from the details of which CPU is actually running the software at any given time and synchronization with other processes on other CPUs is relatively transparent. The application developer can view the system as a uniprocessor environment rather than a distributed processor environment. In addition, the underlying operating systems for SMP architectures are well understood and many robust implementations can be found in the SMP server market. In MPP architectures the application developer must design the application to make use of the distributed processing environment. The MPP architecture must also have a distributed operating system implementation to provide the infrastructure on which applications are layered. Similarly, LCC architectures require applications developers to design applications to take advantage of the distributed processing environment. LCC architectures generally use a nondistributed operating system and place the distributed communications functions at the application level. This provides less support for interprocessor communication than in MPP architectures.

The following is a list of the software architectures that are most commonly mentioned in the video server arena:

- Microsoft Tiger - This software architecture is targeted to an LCC architecture and is based on the Windows NT operating system. The Windows NT operating system is currently ported to the Intel X86 and MIPS processors.

- Silicon Graphics, Inc. / Interactive Digital Solutions - This software architecture was developed for the Time Warner Cable Full Service Network trial in Orlando and is tightly coupled to the Silicon Graphics SMP server architecture [10].

- Oracle Media Server - This software architecture is based on the Oracle relational database management system (RDBMS) and may be ported to a number of different hardware platforms. The article "Oracle Media Server: Providing Consumer Based Interactive Access to Multimedia Data" [11] discusses the Oracle Media Server in detail. This software has been implemented initially on the nCube MPP architecture.

- Sybase Intermedia Server - This software architecture is based on the Sybase RDBMS, although, it proposes to integrate the RDBMS with an independent video server rather than provide the video server software itself.

Impact on Scalability

Scalability is the measure of how the capacity of video server can be increased to support additional video streams. This can be achieved by adding CPUs, network interfaces, disk interfaces, additional servers, or any combination of the preceding elements.

In general, the SMP architecture scales well up to the maximum number of streams supported within one server. Provided the system bus bandwidth is sufficient, additional CPUs, network interfaces, and disk interfaces can be added to the system without requiring software changes. Beyond the limits of one server, additional servers must be added to increase capacity.

The LCC and MPP architectures have the potential to scale to larger number of streams within one video server than SMP architectures. In the SMP architecture you can only add so many CPUs and interfaces to a server before you have to add more servers. MPP and LCC architectures are designed to scale to larger numbers of CPUs and interfaces within one video server.

### Impact on Reliability

Since a video server is a shared resource that may serve thousands of customers, reliability is critical. A single CPU failure or an uncorrectable memory error can bring down the entire video server in an SMP architecture. Also, since the memory and CPUs are tightly-coupled, errors may propagate to other parts of the server. A similar failure in the MPP or LCC architecture will only kill one of the processing units. If the operating system is designed to detect and correct for this problem the video server can continue to function with this loss. While most SMP servers do have the ability to map out failed hardware as the system boots, temporary loss of service still results as the system is rebooted. To accommodate this aspect of SMP architectures, redundant video servers are used. This will dramatically affect reliability and mean-time-between-failures (MTBF) figures.

### Impact on Cost

There are two critical costs to consider when evaluating video servers:

- Entry-level costs
- Incremental-scaling costs

The entry-level cost for SMP architectures is moderate due to the cost of the high-bandwidth, system bus infrastructure. The SMP architecture carries lower incremental expansion cost, however, up to the limit of the server capacity, after which a new server must be added.

The entry-level cost for MPP architectures is fairly high due to the cost of the infrastructure to support the large hypercube interconnect. However, the MPP architecture carries lower incremental expansion cost to scale to larger numbers of video streams.

Finally, the entry-level cost for LCC architectures is low due to the leveraging of standard off-the-shelf component computers. The incremental expansion costs are also low for the same reason.

## ISSUES FOR FUTURE DEVELOPMENT

There are a number issues that require further research and development:

- Use of tertiary storage
- Cost
- Interoperability

The following sections discuss each of these issues in turn.

### Use of Tertiary Storage

The cost of disk storage is a major component of the total video server cost. The use of tertiary storage media such as magnetic tape or CD-ROM can significantly reduce the cost-per-megabyte of storage. Unfortunately, these media do not have the access latency and bandwidth capacities to directly support video delivery. In the future it will be necessary to develop sophisticated video data caching schemes to take advantage of these low-cost storage media.

## Cost

As discussed earlier, entry-level cost and incremental-scaling costs are the important cost factors. Video server costs will continue to be a large portion of the over all end-to-end cost of providing interactive television services. Thus, it is necessary to make the most effective use of the video server capability as is possible. The article "Making a Cost-Effective Video Server" [12] discusses how placement of video data files in either dynamic random access memory (DRAM), disk storage, or tertiary storage affects the cost-effectiveness of a video server implementation.

## Interoperability

There at two types of interoperability that are important. The first is interoperability between video servers and HCTs and the second is interoperability between the servers themselves. In order to make the most cost effective decisions, the network operator must be able to choose the most cost effective settop or server with out regard to interoperability. It is desirable to have the ability to mix and match video servers over time to increase overall capacity and to take advantage of the latest improvements in technology. Standards groups such as DAVIC are working towards defining standards for these levels of interoperability.

## ACKNOWLEDGMENTS

The author would like to acknowledge the tremendous efforts of the Time Warner Cable Denver and Orlando teams in building the Time Warner Cable Full Service Network.

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