Building Resilient Transmission Schemes Over HFC Networks for Telephony

Bouchung Lin, Ramin Borazjani, Scott Hardin

Scientific-Atlanta, Inc.

P.O. Box 6850, ATL 52-D

Norcross, GA 30091-6850

ABSTRACT

Past studies have quantified the behavior of interference and ingress in the reverse path (e.g., 5 - 40 MHz) of Hybrid Fiber Coaxial cable (HFC) plants. Unfortunately, the speculation of a hostile environment has been confirmed. The studies show the detrimental impact of ingress and impulse noise on the transmission quality, and confirm its unpredictable nature.

Preventive measures can be applied to improve cable plants' quality for digital services, but the real challenge is to design a network architecture that survives adverse conditions. Achieving high signal quality is even more important when telephony service is added because it demands high system reliability and fault tolerance.

This paper focuses on the selection of various transmission schemes for this potentially hostile environment. It shows the advantages FDMA (Frequency Division Multiple Access) has over the wideband burst mode TDMA (Time Division Multiple Access). The requirements and mechanism for providing reliable telephone service will be discussed and analyzed. A reliable RF channel protection algorithm and family of channel hopping algorithms that achieve acceptable telephone service will be discussed. System capacity can be estimated given a desired service quality as measured by blocking probability.

INTRODUCTION

As HFC network operators choose to deploy two-way digital services (e.g., telephone service) over HFC, they may consider upgrades to their networks. The extent of these upgrades will be dictated by the digital service equipment's resiliency to ingress. Some vendors' equipment may require a network free of ingress. This may require installation of passive filters at each customer's premises. Other vendors' equipment may employ schemes that readily adapt to ingress that would discount the need for filters at each subscriber's premises. One important factor that affects telephony equipment's ability to adapt to increase in the upstream path is its multiple access method.

Selecting a multiple-access method for upstream resource allocation (a channel between the subscriber and headend equipment) is a most controversial and important decision in building a cable telephony system. Two of the most popular approaches include FDMA and TDMA. FDMA divides the available frequency band into a group of small channels, each dedicated to a telephone line. TDMA has a single carrier that supports several telephone line (typically 24 to 32 lines), with each line assigned a different time slot. FDMA techniques are often used with TDMA systems to juxtapose several TDMA carriers on the same upstream path. The two methods are shown in Figure 1.

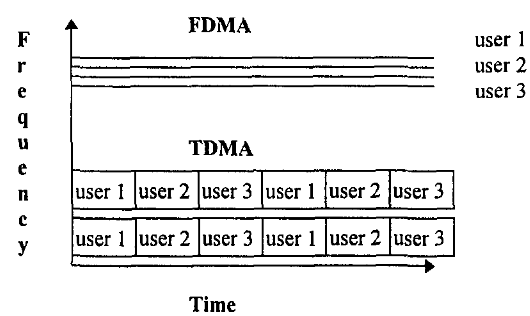

Figure 1: FDMA versus TDMA

FDMA is hypothesized to be more resilient to ingress than TDMA because it can more efficiently be isolated from the contaminated band of spectrum. The first part of this paper quantifies this advantage FDMA systems have over TDMA systems.

In order to help guarantee wireline telephony grade of voice quality, the performance of each upstream channel is monitored by the headend equipment. The system capacity is measured in terms of the number of channels (DS0s) for a given RF spectrum band. A set of channel monitoring algorithms are described in the second part of this paper. A channel hop will be triggered once a corrupted channel is identified. The hopping operation must be brief, and ideally transparent to the end user. Each channel hopping strategy will uniquely impact system capacity, and has a different implementation complexity and algorithm efficiency.

FDMA's ability to monitor individual RF channels (or DS0s) provides an effective mechanism for characterizing and efficiently managing ingress in the upstream RF plant. By dynamically performing upstream spectrum surveillance on narrowband carriers, an HFC network manager can manage applications spectrum usage based on traffic requirements and ingress levels to provide high utilization of the upstream spectrum.

WHY USE A NARROW BAND CARRIER?

One TDMA carrier typcially occupies 40 - 50 times the RF spectrum compared to one FDMA carrier; however, TDMA's channel carrying capacity is significantly less than multiple FDMA carriers occupying the same bandwidth. This is caused by the additional overhead required by burst mode TDMA. Intuitively, an FDMA narrow band system is more efficient in isolating the trouble spots, and efficient at filling gaps between noisy bands due to its smaller channel size. The question remains how much is it better than the TDMA approach? Quantifying this is not an easy task.

Rogers Engineering of Rogers Cablesystems Limited, Ontario, Canada, measured ingress on an HFC reverse plant representative of a fiber to the serving area architecture [3]. Measurements were taken over a period of three days. A review of these field measurements demonstrated that ingress noise typically consumes between 150 and 200 KHz of reverse bandwidth between 5 and 15 MHz. The width of the ingress is significantly smaller than TDMA carrier widths, typically 2 MHz, but is wider than the channel size of the FDMA system, typically 50 KHz or less.

Traditionally, information theory quantifies the quality of the transmission channel by its Signal to Noise Ratio (SNR) under the two conditions: (1) SNR vs. BER (Bit Error Ratio) which is meaningful when considering the same modulation techniques, and (2) as a white noise system. Without dispute, the first assumption is satisfied by assuming QPSK (Quadrature Phase-Shift Keying) modulation for both access methods. The white noise assumption is not satisfied by the ingress patterns observed by Rogers. Becasuse the relative performance is the focus, it is probably not critical to assume white noise for both systems in order to make the comparison possible. However, we will revisit the assumption later.

We segment the return path bandwidth (5 to 35 MHz) into either 600 x 50 KHz (1.5 bits per Hertz to form a 64 Kbps DS0 plus additional overhead) channels for the FDMA system, and into 15 x 2 MHz channels for a TDMA systems as shown in Figure 2. We also assume that the TDMA system uses a signal carrier power 40 times (16dB) stronger than the one used by the FDMA system since we assume that TDMA channel is 40 times wider than FDMA channel.

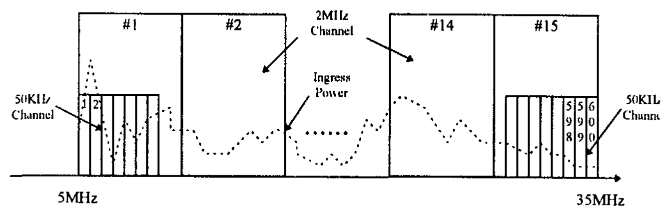

Figure 2: Reverse RF Bandwidth Allocation

Clearly, every channel suffers from different levels of ingress that consequently causes SNR values to vary from channel to channel. Note that the noise power is the total noise power measured within the channel bandwidth. The SNR for each channel is calculated. A distribution of SNR in each system is then defined as $P[SNR_i \leq S]$, which is the probability of the channel having a SNR less than S. The probability distribution functions for both systems is presented in Figure 3.

If the minimum requirements (BER of 10^{-3}) of SNR for QPSK that provides acceptable performance for telephone service is 13 dB, the FDMA system can provide 94.3% of its 600 x 50 KHz channels, or 565.5 channels (one channel carries one telephone line). For the same requirement, the wide channel system can only offer 78% of its 15 x 2 MHz channels, which is 11.7 channels (one channel carries 24 to 32 telephone lines). Therefore, for TDMA these 11.7 channels will provide only 280 to 374 DS0 lines. Thus, for the likely experienced SNR range of 10 dB to 25 dB, the actual efficiency advantage by using the FDMA approach over a TDMA system will be between 50% and 100%.

At low SNRs, the wide channel system delivers slightly better performance than the FDMA system. But QPSK is not normally expected to be deployed in environments with SNR less than 10dB. Even if it is, the difference is very small. As shown in Figure 3, if the reverse channel is exceptionally free of ingress, or when high SNRs are present, there is little difference between these two approaches.

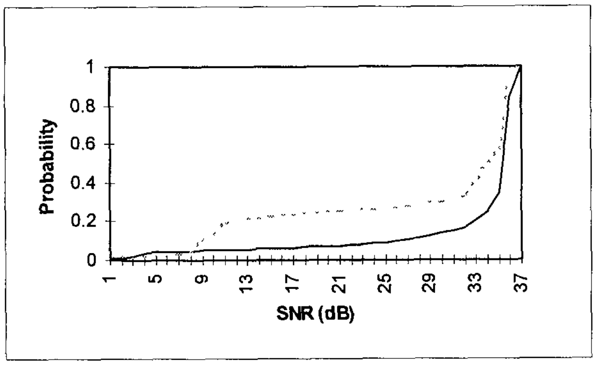

Figure 3: Probability Distribution of Channel SNR for Narrow Band (dashed line) and Wide Band (solid line) Band Carriers over 5 to 35 MHz.

The above comparison is based on the assumption that the transmission performance (bit error ratio) would be the same if the SNR of two band-limited signals of different sizes are identical. This statement is absolutely true only if the noise in these two systems is identical (e.g., white noise).

The data collected by the Rogers engineers demonstrates that the noise characteristics of the same channel change over time, as shown in Figure 4 and Figure 5. The two curves represent the observations at intervals which are three hours apart. In Figure 4, the characteristics of the ingress change very little between the two samples. In Figure 5, taken at different frequencies, the ingress changes dramatically between the two observation times.

These examples also show that ingress either occurs at one or two frequencies, but is unlikely to occur within three or more different frequencies inside of a 2 MHz window. Unfortunately, the resolution of the measurement is not fine enough to show the ingress activity within a 50 KHz band. One can imagine that the ingress is likely to appear as a single interferer, if it is present.

Because the channel noise does not appear to be identical in the two bands (the white noise assumption is violated), the above results are in doubt. Validation of the results is provided in the next section to address this concern.

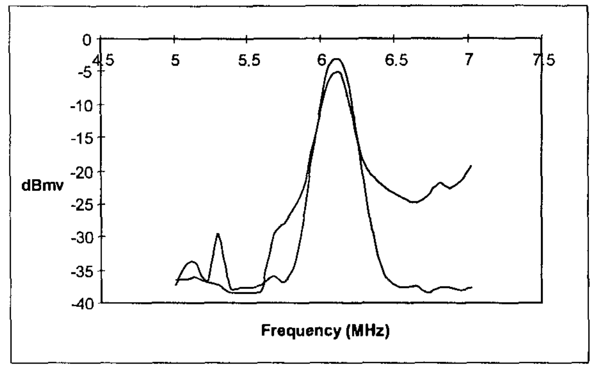

Figure 4: Two samples of Ingress between 5 MHz and 7 MHz

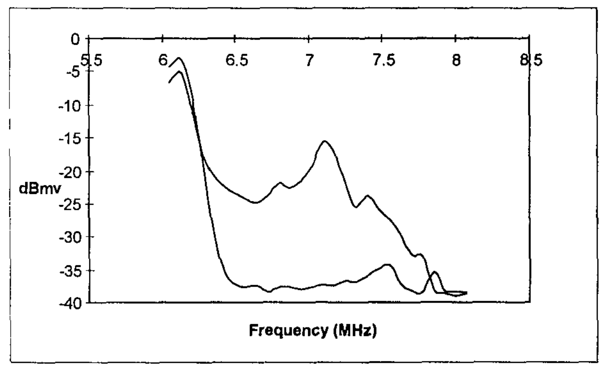

Figure 5: Two Samples of Ingress between 6 MHz and 8 MHz

VALIDATION OF THE RESULTS

A simulation technique is chosen to verify the noise assumption. The simulation model is shown in Figure 6. A stream of digital data is encoded and passed through the QPSK modulator. The modulated signals would be contaminated by the injected noise in the transmission medium. The combined signals (the original signals and the noise) is demodulated. The voltage difference between the output signals and the input signals are collected and compared to the power ratio of the carrier and noise.

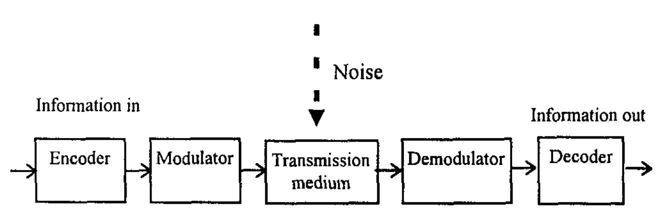

Figure 6: Ingress Simulation Model

Because the objective is to study the QPSK performance under different ingress conditions, we focus on the following two ingress types:

- Ingress, of identical total power, consisting of several different numbers of sinusoidal signals.
- Using the same number of sinusoidal signals located at different frequencies within the QPSK carrier band.

For this purpose, ingress is constructed within a 50 KHz band in four forms as shown in Figure 7. The ingress appears as sinusoidal waves at varying positions from the central frequency: (a) ingress composed of four identical sinusoidal signals at ±1 KHz, ±2 KHz from the carrier's central frequency, (b) ingress composed of four sinusoidal waves at ±16 KHz, ±17 KHz from the central frequency, (c) ingress composed of ten identical sinusoidal waves at ±1, ±2, ±3, ±4, ±5 KHz, (d) ingress composed of ten sinusoidal waves split at ±1, ±2, ±3, ±16K, ±17 KHz, and (e) white noise. Note that the SNRs of each case is fixed, and identical. Even though these waves are modelled in 50 KHz bands, the results will be the same if the frequency is scaled to a 2 MHz band.

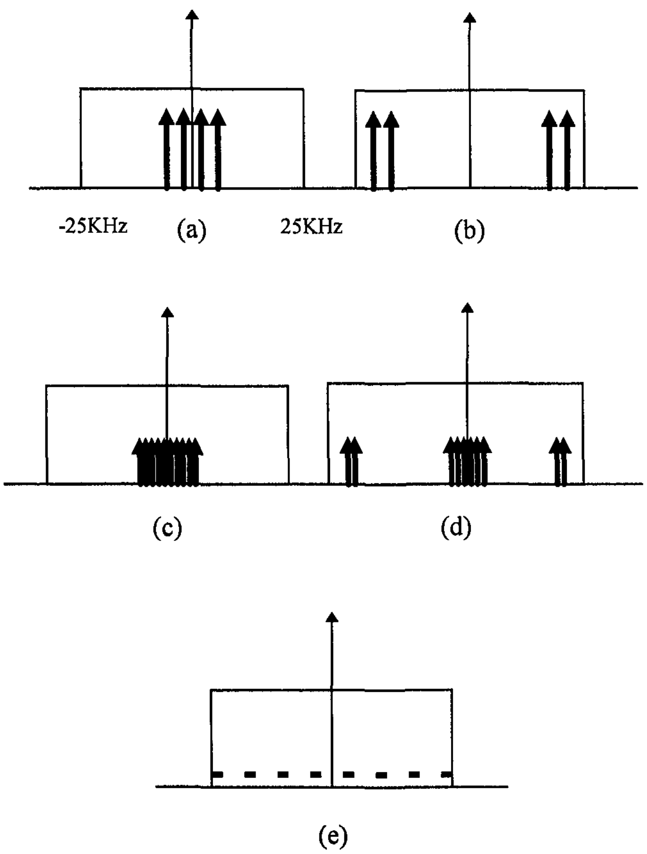

Figure 7: Ingress Models: (a) ±1K and ±2K Hz (b) ±16K and ±17K Hz (c) ±1 to ±5K Hz (d) ±1 to ±3, ±16, ±17 (e) White Noise.

The probability distribution function of each case is presented in Figure 8. The Gaussian white noise has the smallest tail probability (i.e., narrow distribution). Longer tails contribute to higher bit error ratios. The other cases show significantly larger tails, and thus higher BERs compared to white noise. Case d, two groups of ten sinusoidal waves, has the greatest detrimental impact on the performance. Even though the sample space (1888 samples) is not large enough to show the bit error probability, one still can conclude that when the ingress is split into two groups it will generate a longer tail than if it appears as a single interferer.

In all the cases, the total power of the ingress is identical. That is, the power per sinusoidal wave of cases a and b is greater than that of cases c and d. Based on this analysis, several conclusions can be drawn:

- Ingress in two bands within a carrier will increase the error rate compared to a single ingress band.
- Five sinusoidal waves of ingress will cause more errors than two sinusoidal waves.
- Bit error caused by the ingress adjacent to the central frequency is worse than if it occurs away from the central frequency.

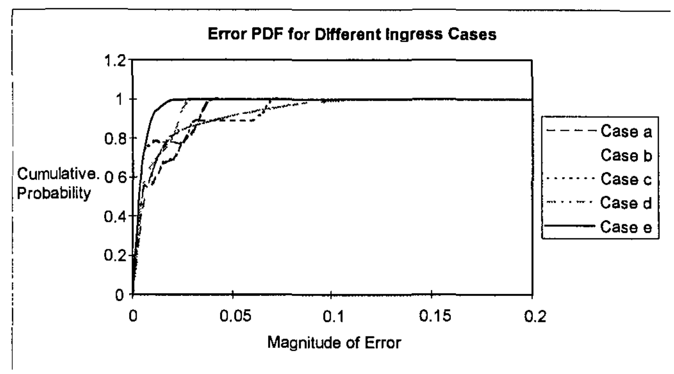

Figure 8: Probability distribution functions of Errors caused by different forms of ingress.

Most importantly, the variation in the magnitude of error is subtle from case to case. This allows us to conclude that with the same SNR, the narrow channel would perform better or at least as well as the wide channel approach. Given that the entire 5 to 35 MHz range is included in the analysis, providing a wide range of ingress scenarios, the approach taken is conservative from an engineering perspective.

RF PROTECTION ALGORITHM

The RF protection algorithm is influenced by the minimum RF channel qualifications for acceptable telephone service [4]. The algorithm includes the thresholds that trigger the frequency hopping mechanism in order to maintain acceptable service. The key prospect of this algorithm is the responsiveness of the mechanism under a given noisy environment.

First, a generic upstream framing structure is presented. The framing is tightly coupled with the RF protection algorithm. The upstream is divided into a sequence of continuous fixed-size frames (or slots). A frame consists a data field carrying DS0 data and a header that consists of an Upstream Frame Alignment (UFA) field, a Cyclic Redundancy Check (CRC) field, and necessary signaling and control information.

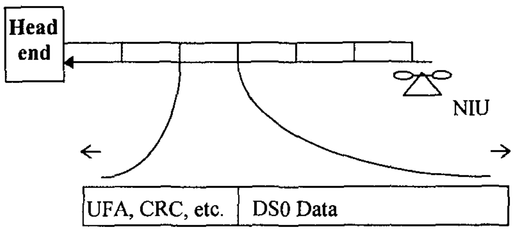

Figure 9: Upstream framing structure

UFA is a fixed bit pattern indicating the beginning of a frame. The CRC field is used to protect the information in the data field at the previous frame.

CHANNEL STATISTICS AND THRESHOLDS

The statistics of the following possible events can be constantly monitored at the headend for each RF channel to measure its transmission quality:

- Code Violation (CV) occurs when a CRC failure is detected.
- Errored Second (ES) is a second in which one or more Code Violations occur [4].
- Out Of Frame (OOF) is the event when headend can not locate the UFA.
- Loss Of Frame (LOF) occurs when OOF persists for a fixed number of times, say L.

There are potentially more events can be defined if implementation is permitted.

The threshold crossing indicates that performance of a channel has reached an unacceptable level. The appropriate threshold values for the above events should be derived from service requirements of POTS and ISDN.

Examples of possible channel-hopping criteria can be:

1. Occurrence of the LOF event, or
2. Occurrence of M CVs in one second.

EXPECTED DETECTION TIME

Given the event definitions, the threshold values, and the system parameters, the probability of these two criteria can be derived.

The derivation of the probability of LOF event $P[LOF]$, for example, starts with the event of OOF. The probability of an OOF is the probability of at least one bit in the UFA field is in error. Given an UFA consists of b_{UFA} bits, the probability $P[LOF]$ is

$$P[LOF] = OOF^L = (1-(BER)^{b_{UFA}})^L \qquad \textbf{EQ. 1}$$

FER (Frame Error Rate), used in the following equations, is defined as the probability of a frame of b_{FRAME} bits containing at least one bit error (a code violation), and is expressed as $FER = 1-(1-BER)^{b_{FRAME}}$.

For the event of more than M code violations in one second , the probability can be expressed by the following equation given there are f frames in one second:

$$P[M\,CV \text{ in one second}] = \sum_{i=M}^{f} \binom{f}{i} FER^i (1-FER)^{f-i} \qquad \text{EQ. 2}$$

The LOF event is designed to deal with the catastrophic condition such as a malfunctioning transmitter. The LOF event would not be expected to occur under typical RF degradation (e.g., thermal noise).

The graphical demonstration of the expected detection times versus BER is given in Figure 10. As the figure suggests, the algorithm will not allow channel hopping to take place if the channel is relatively clean. Using different M, the algorithm would activate the frequency hopping mechanism in one second when BER of the transmission environment exceeds a desired value.

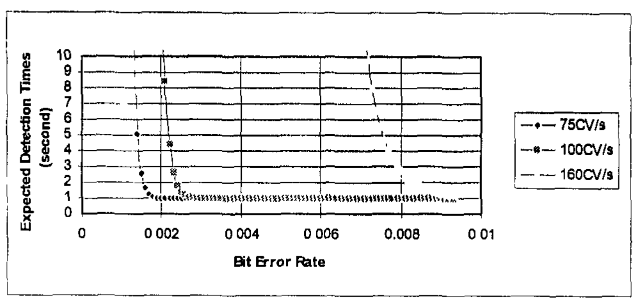

Figure 10: Expected Channel Detection Times

The minimum BER requirements for modem, fax, and telephone modem are not clear at this moment and they may not be as tolerant as the voice application. If a lower

BER is required, the algorithm can be tuned by modifying the parameters.

UPSTREAM CHANNEL ALLOCATION ALGORITHMS

So far, we defined the channel qualification and the channel hopping criteria. Consequently, a channel allocation algorithm is needed in order to choose a new channel when the current channel is declared unusable. The variations of such operation will dramatically affect the hardware and software complexity. A system may choose to implement a subset of the following functionality which may result in different system efficiency. The possible system features are:

1. *Assignment of RF channels to subscribers*: Fixed or Dynamic. *Fixed* means that a channel is assigned to a telephone line permanently. The relationship still exists even when the telephone set is in an on-hook state. *Dynamic* implies that the RF channel is released to a channel pool once the telephone set is on-hook. An off-hook event will cause the headend equipment to assign a channel. The mechanism the Network Interface Unit (NIU) uses to request a channel upon a off-hook event is beyond the scope of this paper, and will not be discussed.
2. Assignment of an NIU to a Demodulator / Receiver: Fixed or Variable. In practice, several Demodulators / Receivers can be grouped together. The NIU may be able to tune to only one group (fixed), or more than one group (variable).
3. Channel Performance Selection Classification: None or Queue. Once a channel is determined to be released due to the ingress noise, an abnormal release, it can be treated differently. A Queue, or linked list, can be maintained at headend such that a normally released channel can be enqueued at the top of the list and a abnormally released channel is enqueued at the end of the list. New calls would be established with preference for the channels that were released normally.

Different combinations of above functionality would result in different system performance with respect to the call blocking probability. For a given requirement, the results would be used to determine the maximum RF channel concentration ratio, and the system capacity.

SYSTEM QUALITY OF SERVICES (BLOCKING PROBABILITY) STUDY

The performance of a telephone system is traditionally described by the blocking probability of a call attempt [1]. To demonstrate how system capacity can vary due to different allocation schemes, we briefly review the queueing models commonly used in the telephony system.

THE EXISTING SYSTEM

A telephone system with m DS0 circuits is usually described either as an M/M/m or an M/M/m/n system, as shown in Figure 1, depending on whether calls are generated from an infinite number of phone lines or a finite number, n, of phone lines. The arrival pattern is a Poisson process. The length of call holding time is exponentially distributed with mean time, t.

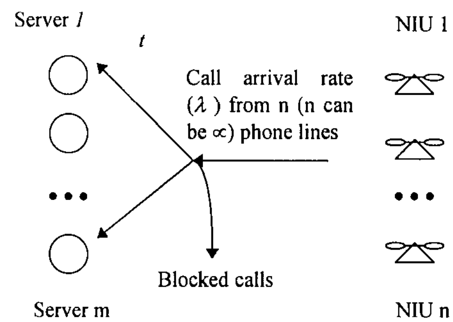

Figure 11: M/M/m/n Queueing Model

The blocking probability of a M/M/m system is known as the *Erlang's loss formula or Erlang's B formula:*

$$B_{infinite}(m,\lambda t) = \frac{(\lambda t)^m / m!}{\sum_{k=0}^{m} (\lambda t)^k / k!} \qquad \textbf{EQ. 3}$$

where m is the number of servers, λ is the average call arrival rate, and t is the mean holding time [2]. In a real system, the assumption of infinite customers is not realistic if the RF concentration factor is not greater than five. The above system has been further modified such that calls are generated by a finite number of telephone lines and the blocked customers will not generate another request immediately, but will after an exponentially distributed idle time. The blocking probability for such system is:

$$B_{finite}(n,m,\lambda t) = \frac{\binom{n-1}{m}(\lambda' t)^m}{\sum_{i=0}^{m}\binom{n-1}{i}(\lambda' t)^i} \qquad \textbf{EQ. 4}$$

where n is the number of customers, m is the number of servers and λ' is the call generating rate for one idle telephone line, and t is the average holding time for a call [2]. One can argue that this is not realistic either because a blocked customer may very well generate another request right after being denied by the system. Nevertheless, Equation 4 can be viewed as the upper bound, and

Equation 5 the lower bound of a real system. The difference of these two systems will not be significant once the concentration factor is greater than five.

OUR MODEL

In a HFC environment, the channel impairment can temporarily disable a channel for a undetermined period of time that makes the number of servers become a stochastic random variable.

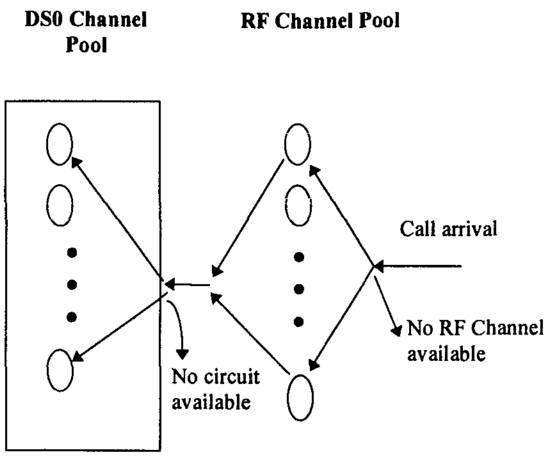

Figure 12: Queueing Model

A call can be rejected by either the DS0 circuit being unavailable or RF channel being unavailable, but not because of both events. One can conclude that the overall system blocking probability would be determined by the more constraining event. Since the two blocking events are mutually exclusive, one can further decompose the system into two parts: one consisting of the DS0 circuit pool (limited by switch ports) as the servers, and the RF channel pool (limited by the available RF channels) as the servers.

Recall the observation from the previous section, a HFC plant may lose a number of RF channels due to the RF impairment. But the stochastic behavior of the effective number of RF channels is not well known, and probably varies from plant to plant. It is not practical to manage the system capacity based on reliable assumption. The conservative approach is to provision a number of spare channels to accommodate channel impairments to maintain the desired service quality based on the worst scenario. The rest of this section will focus on the channel assignment algorithm.

CALL BLOCKING PROBABILITY

Two channel hopping mechanisms are used to demonstrate how the system capacity can be affected. The option of channel selection using a linked list is not included in the study.

OPTION 1

Under Option 1, each NIU transmitter is assigned to one channel. The association will not be released even the NIU is on-hook. If a channel is impaired, the headend equipment will direct the NIU to a spare channel within the same receiver. Therefore, the incoming calls can only be generated from a finite number of customers which is equal to or smaller than the number of channels that a receiver can support. Since the RF channel and NIU transmitter are tied together, the blocking model is slightly different from the M/M/m/n model, but we still can adapt the results to our system.

One observation of Option 1 is that once a NIU is assigned to a RF channel, its quality of service is somewhat determined. A NIU associated with a clean RF channel would never be blocked due to the RF channel unavailability. Blocking only applies to those whose RF channels happen to be in the area that ingress or other sources of impairment are strong. Blocking occurs when the number of impaired channels is greater than the number of spare channels. The blocking probability is no longer identical among the telephone lines. An average blocking probability would not appropriately represent the system blocking characteristics. The customers that happen to be assigned to the impaired channels must be satisfied.

The problem can be stated as follows: given a number of impaired RF channels, how many spare channels are needed such that the traffic generated by the telephone lines assigned to these RF channels will experience a desired blocking probability. Such system can be described by a M/M/m/n model, where m is the number of impaired channels and n is the number of spare channels.

The three-dimensional diagram in Figure 13 is based on a 24 channel receiver, and demonstrates the relationships among the following parameters: the number of impaired channels, the number of spare channels, and traffic density. Four different traffic levels, CCS equal to 3, 6, 9, 12, are considered. Note that the z axis is inverted for readability. The results confirm that one spare channel is required for every expected impaired channel when the total number of impaired RF channels is less than or equal to five channels in order to maintain the requirement of 1% blocking probability. If the HFC plant is likely to have more than five channels impaired, fewer than one for one spare channels will be needed because of the unlikelihood of all spare channels being needed simultaneously.

for one spare channels will be needed because of the unlikelihood of all spare channels being needed simultaneously.

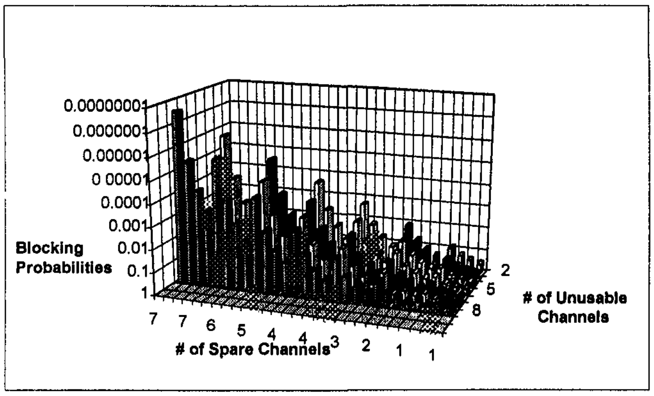

Figure 13: Blocking Probability for a Non-Concentrating System

The service providers need to characterize their reverse channel characteristics in order to determine the number of spare channels if Option 1 is implemented.

Using the results from the previous section, 6% of RF channels will be impaired, two out of 24 will be set aside. Therefore, 550 phone lines can be supported for 5 MHz to 35 MHz bandwidth.

OPTION 2

The above system can be improved by dynamically assigning RF channels between the NIU and headend equipment. An RF channel is assigned to an NIU when it is needed. Each channel will be returned to the channel pool after the call is terminated. Therefore, a telephone line can be assigned to any RF channel within the same receiver. No designated spare channels are needed. This scheme allows an impaired user to use any free channel to improve the channel utilization.

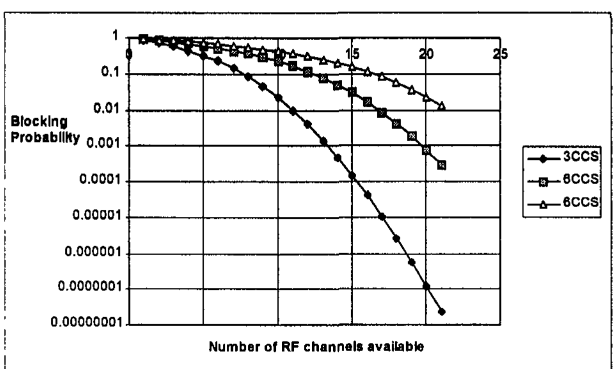

Figure 14: Blocking Probability of a 70 phone-line system

Clearly, the system blocking probability is determined by the HFC system characteristics. Given a finite number of phone lines the blocking probability versus the number of occupied channels can be provided based on Equation 5, as shown in Figure 14. Using the unavailability rate of 4%, a 24 channel receiver can support up to 70 phone lines. Comparing this result to the 20 phone lines that the same capacity receiver supports under Option 1, a more than 200% gain is achieved.

NETWORK MANAGEMENT IMPLICATIONS

If network operators implement systems that use narrow upstream carriers (50 KHz), the benefits can be extended with HFC network managers that perform upstream spectrum surveillance and management. The network manager can use the processed performance monitoring information provided by the receiver / demodulators to maximize equipments utilization of the RF spectrum.

The agent consists of equipment located in each headend that measures , at 50 KHz intervals, RF power levels, or Bit Error Rates of signals generated by transmitters located deep in the HFC subnetworks. Initially, the information provided by the network manager could be used to provide assistance in optimally assigning upstream frequencies to data and telephony applications. Later, a standard interface will allow operators to further optimize the frequency assignment by dynamically allocating bandwidth as dictated by traffic. Using a spectrum manager and agent to dynamically measure and assign bandwidth for each upstream HFC subnetwork provides the following benefits.

- Increases service reliability achieved through optimal frequency utilization
- Maximizes digital services' upstream spectrum utilization for each HFC subnetwork
- Allows performance monitoring of the entire upstream spectrum to help identify marginal HFC subnetworks
- Expediently identifies ingress and isolates it to a specific upstream HFC subnetwork

References

1. "Digital Telephony and Network Integration", Bernhard E. Keiser and Eugene Strange, 2nd edition, International Thomson Publishing Inc.

2. "Queueing System Vol. 1 Theory", Leonard Kleinrock, John Wiley & Sons, Inc.

3. "Two-Way Cable Television System Characterization", CableLabs, Inc., April 12, 1995.

4. TR-NWT-000499, Nov. 1991.