instructor (or his aids). The hardware set-up shown in Figure 14 allows the instructor to choose from a very large variety of presentation tools.

In its simplest form, an instructor using vu-graphs and having no knowledge of computers could use the scanner to transmit each individual vu-graph. The sketch-pad or mouse could be used as a pointer or highlighter. Alternatively, the sketch-pad could be used in the same manner as a blackboard is normally used in the classroom.

Hand-out documents could be transmitted for individual distribution on floppy discs or for generating print-outs in the classroom. These documents could be scanned off-line before the class or could be sent as a word processing document file if originally generated on a computer. Thus, all functions normally performed now in a typical classroom could be done remotely and economically.

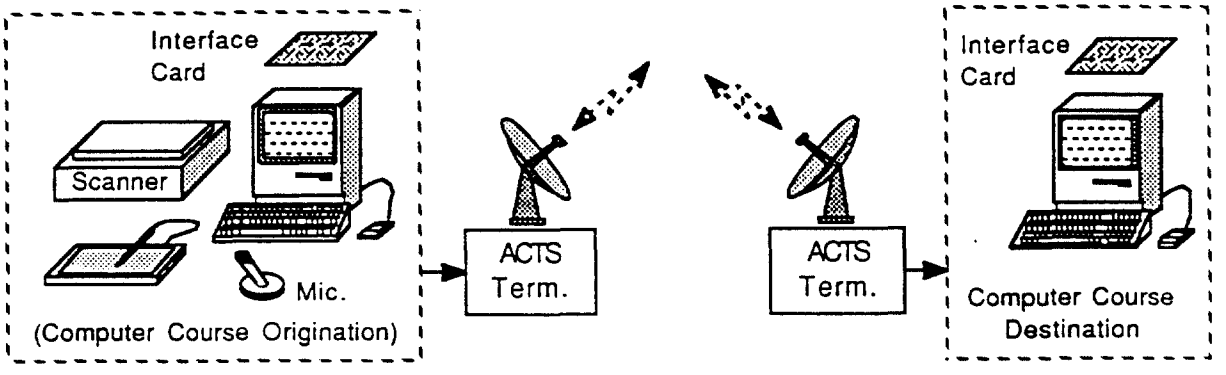

Figure 14 Computer Lecture Course

Most vu-graphs today are generated on a computer and then printed on paper or transparency. However, software is now available (e.g., Microsoft PowerPoint) to do an entire vu-graph presentation directly from a computer (e.g., through the computer screen or an LCD projector) without ever having to print the output. This is ideal for this application.

Many other tools, such as VideoWorks II and Hypercard, are also available today for producing highly stimulating audio-visual animation of presentations. In fact, Hypercard can launch VideoWorks II presentation strips. Needless to say, these tools will continue to improve and new tools will continually become available in the future. Down-loading of the presentation software from the originating source allows these tools to be used as they become available. Part of the beauty of this system is that it will never become obsolete.

Two developments are necessary to implement the described system. One is the hardware interface card to allow transmission of information from computer to computer through the

transmission medium (e.g., a multiple of 64 Kb/s). The other is the control-level software. Some of the software functions that will need to be provided are:

- Control of transmissions
- Coordination of originating and receiving computers
- Management of other available software resources
- Control of transmission resources (e.g., originate and terminate service)

Voice transmission could be provided by using a standard telephone-type PCM voice codec and assigning a dedicated 64 Kb/s transmission capacity for this purpose. Unfortunately (and obviously), that will sound like a telephone call because the bandwidth is only about 3 KHz. Using the same 64 Kb/s capacity, ADPCM coding will allow high-quality voice transmission with about a 7 KHz bandwidth and will sound very natural.

The additional transmission capacity required for the computer/video transmission will probably be in the range of 64 Kb/s to 192 Kb/s for a total transmission rate of between 128 Kb/s and 256 Kb/s. A single 64 Kb/s voice channel should suffice for the return link from the remote classroom(s).

Some applications may require that the transmission bit-rate allocation be changed during a class broadcast. For example, a music theory class might use the same format described above for part of the class but occasionally request 1.5 Mb/s capacity for transmitting excerpts of a compact audio disc to demonstrate the application of theory to practice.

The origination end requires a normal two-way ACTS terminal. The destination end could be either a normal ACTS terminal or a less expensive receive-only terminal. Of course, the latter requires that any return link be provided by other means (e.g., telephone or ISDN data).

Multiple Service with Receive-Only Terminal (most complex)

The most general ACTS experiment for educational broadcasting would be a combined dynamically multiplexed demonstration using new receive-only terminals at different locations as shown in Figure 15. The compressed video could use modified commercially available equipment or the new more-general design discussed elsewhere in this paper. Simulated traffic would provide multiplexer loading that is statistically similar to that expected in an actual operating system.

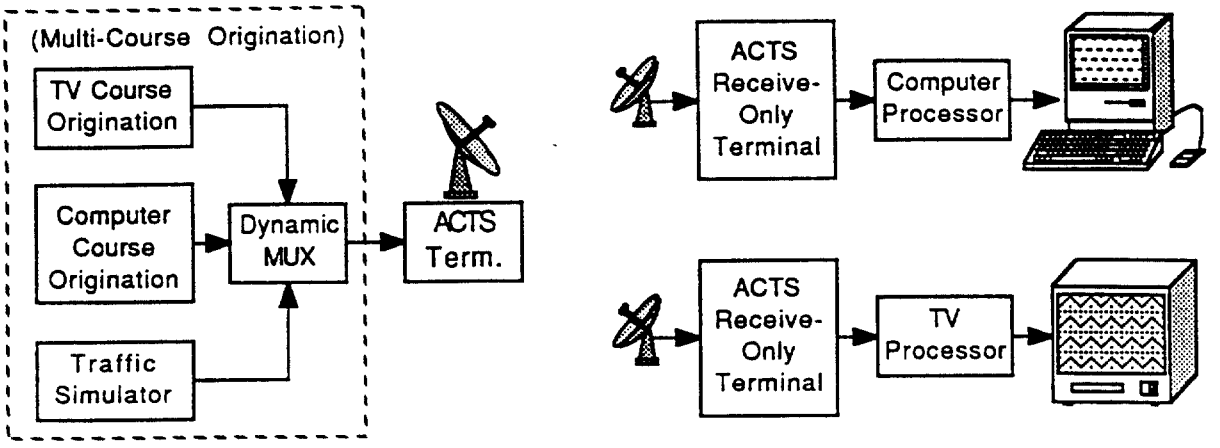

Figure 15 Multi-Service with Receive-Only Terminals

A standard ACTS terminal would be used for origination of services. The two or more remote terminals would use the new receive-only design which would be considerably more economical. Because the destination terminals are receive-only, any return link would have to be by other means such as telephone or data link.

Recommendation

The proposed ACTS experiment is actually a modification of the single-service computer lecture. Because of the high interest in receive-only terminals, it is proposed that a standard ACTS terminal be used initially in a receive-only mode and be operated in accordance with procedures and signal formats that would be used in a manufactured receive-only terminal. This may require modifications to the ACTS control software and to the ACTS terminal software. However, it would be the least expensive approach to proving out the concept of using ACTS as an educational broadcasting system.

Cost/Schedule of Prototype Design

Single Service Computer

The recommended approach for the next phase is to develop the Computer Lecture service using (software) modified ACTS terminals. A task schedule for such a development is shown in Figure 16 and extends over a period of about 10 months. A starting date of October 3, 1988 is assumed, also for discussion purposes.

This is a first cut and is intended mainly for discussion purposes. It may be overly optimistic. Also, it is obviously unrealistic in the sense that real ACTS hardware may not be available during this time-frame. The main value in this exercise is in getting an initial estimate of the development cost and schedule.

The subtask for modification of ACTS terminal and MCS software is shown in Figure 17. The required modifications have been enumerated previously. No cost is included for this subtask (except System Engineering monitoring) because its implementation is beyond our control and must be done by others.

This schedule also assumes that cooperation is received from NASA and the Terminal and MCS contractors in the ACTS system software modifications and in providing the necessary access and GFE.

The direct labor cost estimate is based on the scheduled man-power loading which is approximately one each of a Sr. Systems Engineer, Digital Design Engineer, Electronics Technician, and a programmer. The estimated costs are as follows:

Direct Labor		$ 178 K
Overhead	90%	$ 160
ODC (Travel, Lab equip. lease, etc.)		$ 30
Purchased Equip. and Services		$ 20
Purchasing Burden	10%	$ 2
G&A	20%	$ 78
Profit	10%	$ 52
Total		$ 520 K

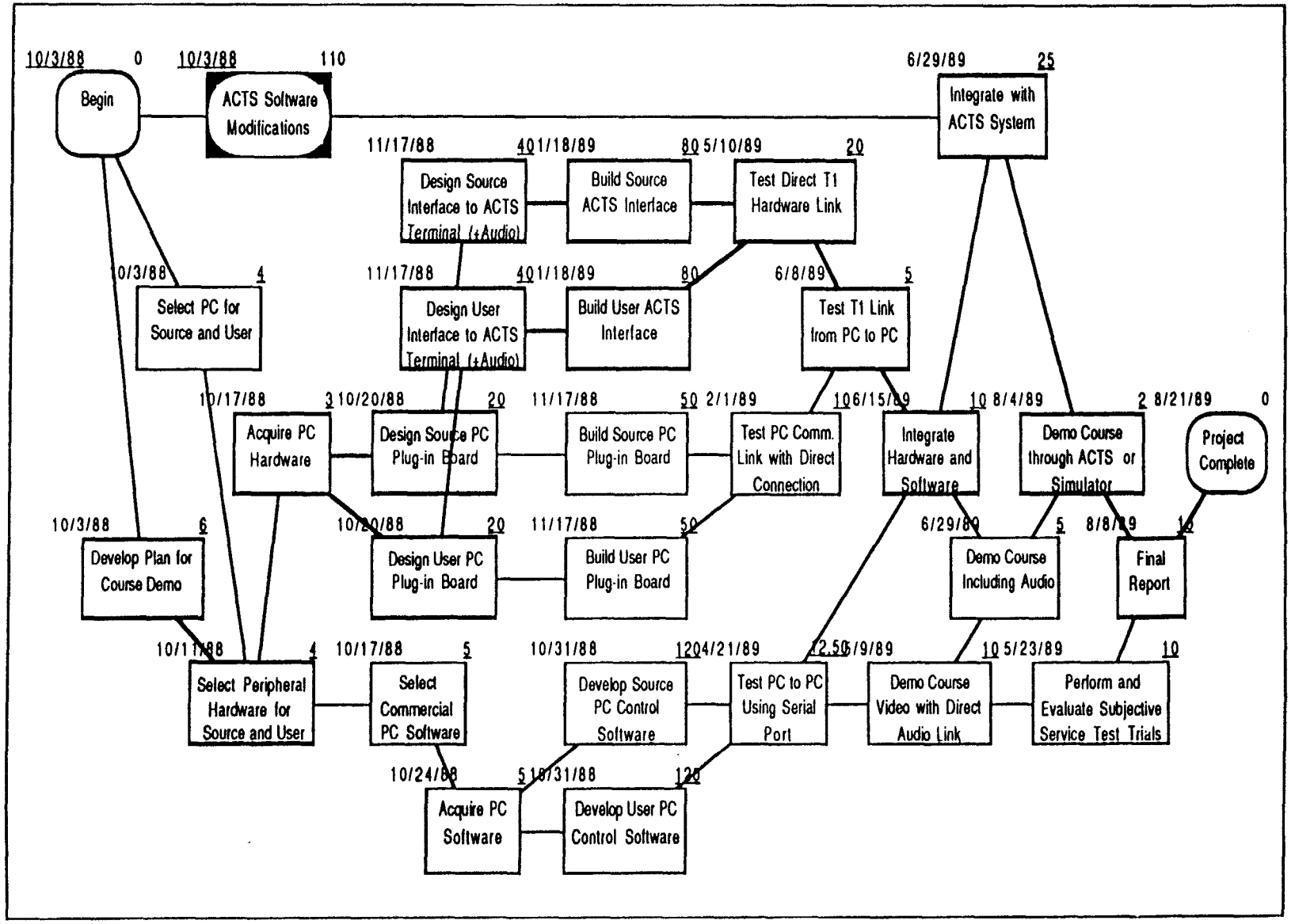

Figure 16 Task Schedule for Prototype System

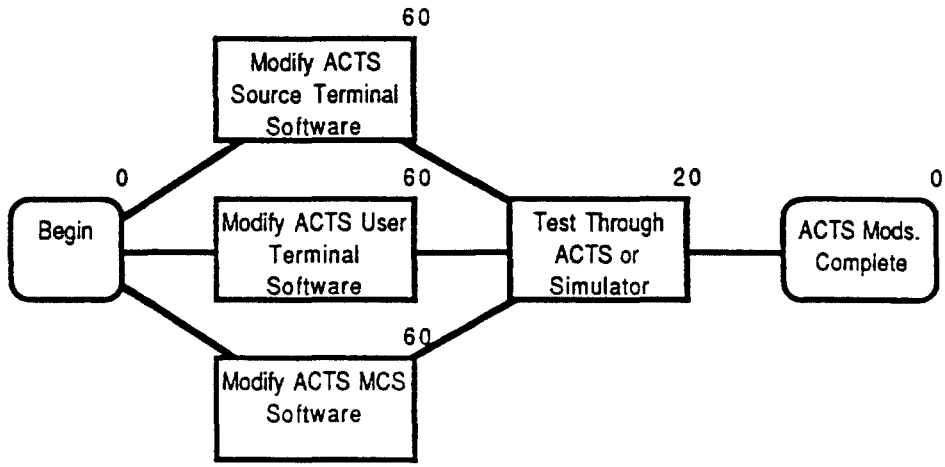

Figure 17 Subtasks Impacting ACTS System

Receive-Only Terminal

A modified Task Schedule is shown in Figure 18 for developing a prototype system which includes the receive-only terminal discussed previously. The actual receiver development is shown as a subtask which is expanded in Figure 19. The duration of this development schedule is about 16 months. Again, this may be overly optimistic.

For estimation purposes, most of the circuit construction is assumed to be subcontracted out at a cost of $150,000. Another $50,000 was allowed for additional hardware such as an antenna dish, LNA, high-stability frequency reference, power supplies, etc. Neither of these figures has been verified with potential vendors and therefore should be considered to be WAGs.

The estimated cost, based on the individual tasks manpower loading is as follows:

Direct Labor		$ 440 K
Overhead	90%	$ 396
ODC (Travel, Lab equip. lease, etc.)		$ 50
Purchased Equip. and Services		$ 220
Purchasing Burden	10%	$ 22
G&A	20%	$ 226
Profit	10%	$ 150
Total		$ 1504 K

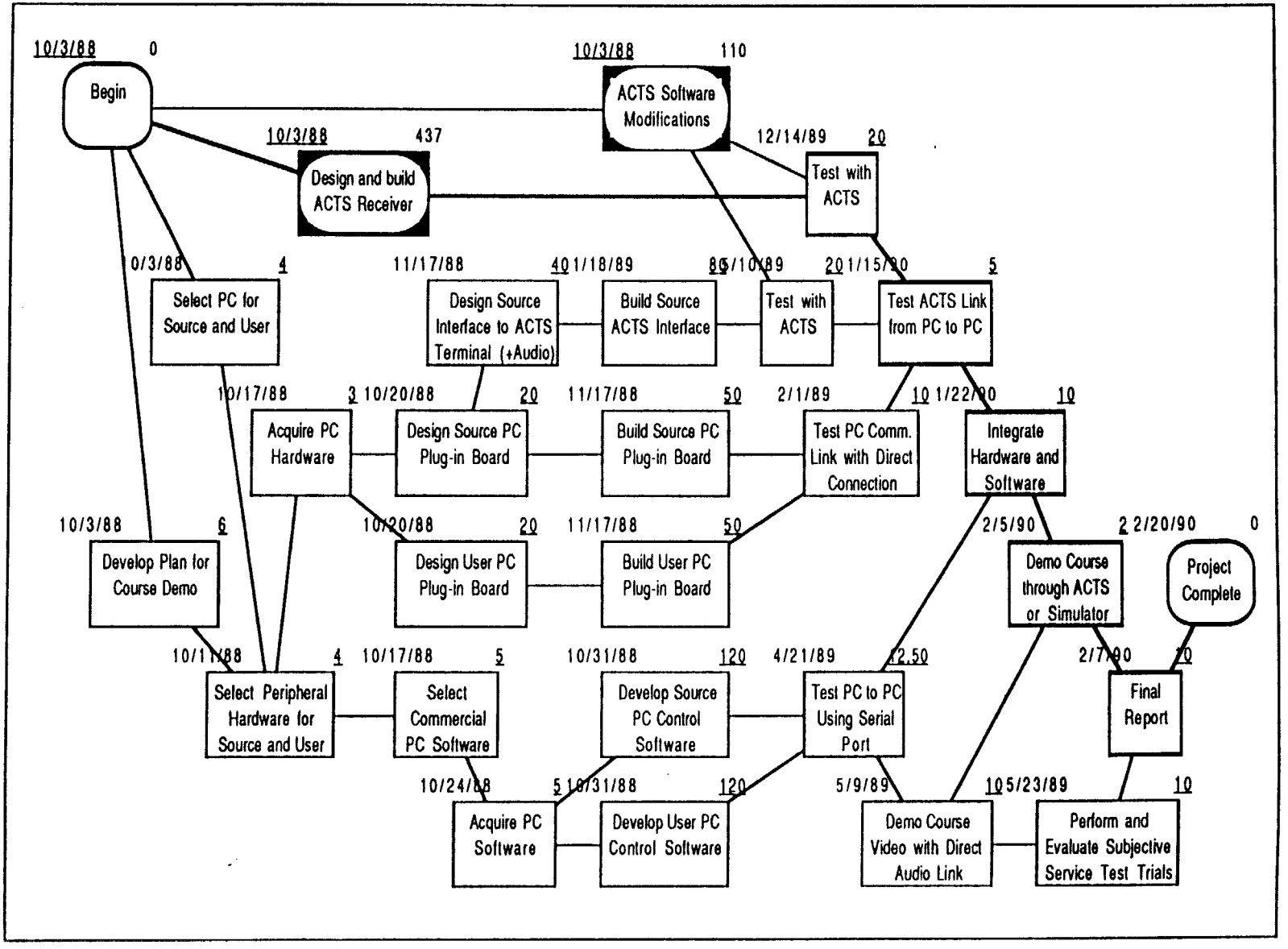

Figure 18 Task Schedule for Prototype with Receiver

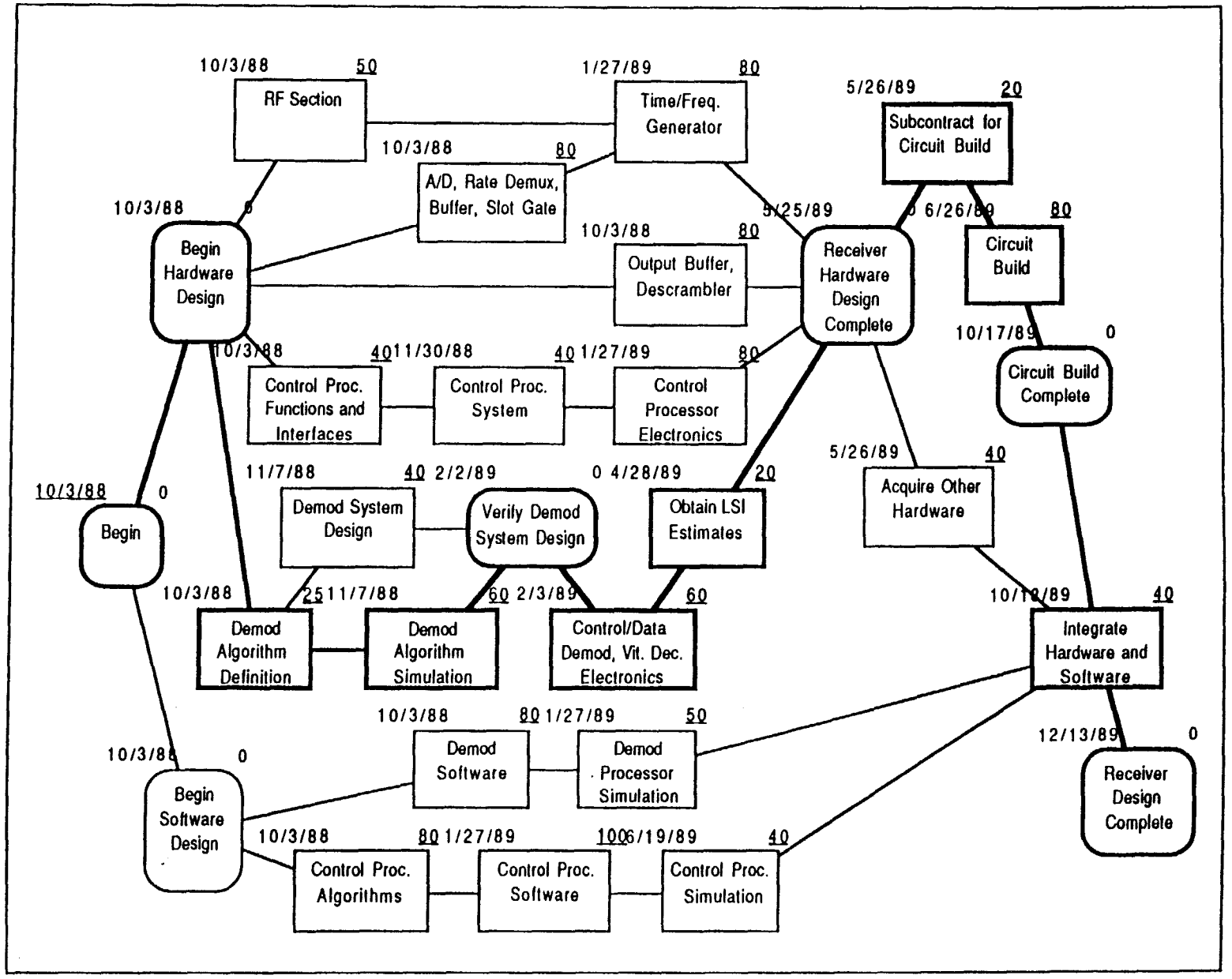

Figure 19 Subtasks Schedule for ACTS Receiver

Conclusion

The ACTS system is a uniquely viable medium for an educational broadcasting service. It is particularly useful for efficient broadcasting to selective areas where local area, locally generated program content is desirable. It also can be used for selective, remotely generated course distribution. However, to accomplish this, certain minimal modifications are required in the ACTS terminal control software and the MCS control software. These modifications are described herein.

A course program format is described which uses personal computers as an integral part of the delivery medium. This is a very efficient means of transmitting very sophisticated educational course programming. By using dynamic multiplexing techniques for multiple course transmissions, even more efficient delivery can be attained. (Analytical techniques for evaluating this are included in Appendix B.) The computer course approach is recommended for an ACTS experiment.

Two development program options for a prototype system are described along with cost estimates, schedule duration estimates and detailed task-schedule breakdowns. One approach uses software-modified LBR ACTS terminals which would be GFE (Government Furnished Equipment). The other includes development of one prototype of a receive-only ACTS terminal designed for a low-bit-rate service of the type described above and using an innovative demodulation technique. The approximate costs and schedule durations for both options are estimated as follows:

Option 1 (Using modified LBR terminals):	≈ $ 500 K	≈ 1 year
Option 2 (Including receive-only terminal):	≈ $ 1500 K	≈ 1 1/2 year

Because the task schedule is heavily front-loaded with system engineering, it is recommended that some of this work (e.g., algorithm development and demodulator simulation) be done beforehand.

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1 G. Gordon Apple, "Digital DBS Educational Services: A Plan for the 1990s," Advanced Communications Corporation, July 31, 1987.

2 G. Gordon Apple. "Proposal to the Public Service Satellite Consortium and the NASA ACTS Program," Advanced Communications Engineering, Inc., October 21, 1987.

3 G. Gordon Apple, "Final Report: High Definition Television Data Compression; Algorithm Analysis and System Design," TRW, June, 1982, prepared for CBS, Inc.

4 G. Gordon Apple, "Digital DBS Television Transmission," Advanced Communications Corporation, June 5, 1987.

5 "Tandy Developes a Compact Disc that's Erasable," *Los Angeles Times*, April 22, 1988.

6 Naderi, F.M., and S.J. Campanella, "NASA's Advanced Communications Technology Satellite (ACTS): An Overview of the Satellite, the Network, and the Underlying Technologies," Proceedings of the 12th International AIAA Communication Satellite Systems Conference, March 13-17, 1988, Crystal City, VA

7 Coney, T.A., T.R. Dobyns, D.M. Chitre and R. Lindstrom, "Service Offerings and Interface for the ACTS Network of Ground Stations," Proceedings of the 12th International AIAA Communication Satellite Systems Conference, March 13-17, 1988, Crystal City, VA

8 NASA Lewis Research Center, Cleveland, Ohio, "Exhibit "A", Statement of Work, LBR Earth Stations for the NASA Advanced Communications Satellite (ACTS) Program," April 1, 1988.

Appendix A - Efficient Compressed TV Transmission

Why digital transmission?

Technology in digital transmission systems has made steady progress since Reeves[1] invented PCM (Pulse Code Modulation) in 1938. The Bell Labs invention of the transistor in 1947 was followed in the '50s with the development and implementation of the T1 carrier system (1.544 Mbps). This system uses PCM to multiplex 24 voice channels over a twisted pair of wires. More importantly, this allowed, for the first time, essentially distortionless regeneration of the signal regardless of distance. This was followed in later years by a family of digital wire, radio, and fiber optic carrier systems and digital switching systems.

In the late '60s and early '70s, Picturephone was developed by Bell Labs. Although it was a commercial failure, significant technological achievements were accomplished. It was early recognized that digital transmission was an absolute necessity for long-haul Picturephone transmission. For this purpose, a DPCM codec was developed which operated at the (also ill-fated) T2 carrier rate (6.312 Mbps). With this codec, only 96 telephone calls were displaced rather than the 300 or so with analog transmission.

Later data compression techniques, using (among others) frame stores and conditional replenishment[2], allowed transmission below the T1 rate thereby displacing less than 24 telephone calls. This was the beginning of commercial video data compression and digital video transmission. Even later, new companies would develop full NTSC-resolution conferencing systems that would displace as little as a single digital telephone channel.

The advantages of digital transmission have long been recognized. However, the increased bandwidth required by PCM encoding has prevented its use on bandwidth-limited channels such as those used in broadcast and satellite distribution. As a result, digital transmission of TV requires video data compression. The above mentioned techniques work well, but have been prohibitively expensive for use in other than shared systems (e.g., common carrier).

Today's VLSI technology and that projected for the early '90s has radically changed the picture in several ways. The frame store that once required a shelf of memory cards can now be implemented with three 1-Mbit chips. Single-chip high-resolution LSI video processors are available today for personal computer graphics. LSI video data decompression for television is just as feasible. A single chip field store (4-bits wide) is available now and a single chip (8-bit wide) frame store is likely soon.

The advantages of digital transmission also include flexibility and future system compatibility. Receivers and video processors can be programmed by periodic down-loading of algorithms directly from the satellite. Compatible systems for High Definition Television (using one or two standard channels) would be implemented by down-loading new algorithms to both standard resolution and HDTV receivers.

Radio and TV audio channels, having the quality of Compact Disk, would be included along with a multitude of digital data services. The latter would be transmitted without specific allocation of channels because of the variable data rate used in compressed video transmission. For example, several educational channels with voice and still-frames or electronic blackboards could be provided simultaneously in a fully operative DBS TV channel without degrading the primary TV signal occupying that channel.

Programmable processors in the digital receivers would allow virtually unlimited flexibility in the mix of services. During one time period, a single channel could simultaneously transmit several full video (limited motion) educational programs (e.g., class lectures) plus other services. During another period, the same channel could transmit a full motion, standard television program (plus other services previously mentioned).

Subscriber access control for each service could be accomplished by totally digital, "hard" encryption of both video and audio. The high transmission data rate of digital DBS allows rapid real-time updates of the subscriber access list. Pay-Per- View services could easily be offered, including such options as parental-discretion lockout.

The immense flexibility of digital transmission also extends into other realms. Many of the educational and other services could be provided at lower transmission rates through flat panel receiving antennas as small as one foot, or even as small as six inches (about the limit for orbital slot discrimination). Again, these services would coexist with other higher-rate services in any desired reassignable mix.

In the remaining sections of this attachment, we examine the transmission channel and the required data compression techniques.

The Transmission Parameters

The first step in the transmission system conceptual design is to determine a feasible data rate that can be economically implemented. Because compression has its limits similar to bandwidth availability, the higher the data rate, the less complex data compression is required.

An earlier study, by TRW[3] under contract to CBS Television to define a method for digital HDTV (High Definition Television) for DBS, found that 40 Mbps per (24 MHz) DBS channel was feasible. Some of the relevant system parameters were:

- 230-Watt satellite transmitter (TWT)
- Quarter-CONUS (time-zone) coverage
- 0.9 m (or 1.1 m) receiver dish size
- 6 channels per satellite

Since that time, analog DBS designs have been proposed with these parameters:

- 125-Watt TWT
- Half-CONUS coverage
- 0.6 m receiver dish size
- 16 channels per satellite

Spreading a smaller amount of signal power over a larger area and using smaller receiver dishes could have decreased the potential for digital transmission. However, during the same time period, other compensating technological changes have occurred:

- User and satellite antenna efficiencies are better
- Receiver noise figures are much better
- Powerful error correcting decoders are now available
- Data compression techniques have improved
  (i.e., lower bit-rate possible for same quality)

In addition, the system presently under consideration is a conventional NTSC-compatible television signal, which requires only 20% of the bandwidth of an HDTV signal (which would have used two DBS channels).

After considering all of these factors, a transmission rate of 30 Mbps was selected for a DBS system. The supporting link budget is shown in Figure A1 and discussion of the individual elements of the table follows.

Figure A1 DBS Digital Link Budget

LINK ELEMENT	COMMENTS	CENTER	Faded/EOC	UNITS
HPA Output power @ EOL	125 Watts	21.0	21.0	dBw
Feed and Diplexer Loss		-2.6	-2.6	dB
Antenna Gain	8.5 square deg.	36.9	36.9	dB
Sat. Antenna Efficiency	70%	-1.5	-1.5	dB
Beam Edge Loss		0.0	-2.0	dB
		---	---	
EIRP		53.7	51.7	dBw
Space Loss		206.0	206.0	dB
Atmospheric Propagation Loss	(Incl. Rain)	0.7	3.2	dB
Pointing Loss		0.5	0.5	dB
Depolarization Loss	(Rain)	0.0	0.5	dB
		---	---	
PATH LOSS		207.2	210.2	dB
Rec. Ant. Gain	0.6 Meter 12.45 GHz	37.9	37.9	dB
Rec. Ant. Efficiency	75%	-1.2	-1.2	dB
		---	---	
Total Receiver Antenna Gain		36.6	36.6	dB
Rec. Noise Temp	1.5 dB Noise Figure	120	120	deg.
Sky Temp.		20	20	deg.
Background Temp.		10	10	deg.
Atmospheric Noise Temp.	(Rain)	0	50	deg.
		---	---	
Total Temp.		150	200	deg.
		21.8	23.0	dB-K
Terminal G/T		14.9	13.6	dB/K
Boltzmann's Constant		-228.6	-228.6	dBw/K-Hz
C/No		89.9	83.7	dB/Hz
Eb/No	30 Mbps	15.2	8.9	dB
Required Eb/No for BER=10^-10 Using 7/8 Rate R-S Code and Optimum Mod.		7.5	7.5	dB
Margin		7.7	1.4	dB

EIRP

The 125 Watt satellite output power is based on a design for a 16 channel satellite. Feed and diplexer loss is 2.6 dB. This is a considerable thermal dissipation design requirement, but one which has been accommodated by manufacturers. Antenna gain is based on the solid angle for a shaped beam with a half-CONUS coverage from a geosynchronous orbital location with near- Northerly aiming point. Multiple feeds allow this power to be almost evenly spread over the coverage area, with rapid fall-off and very low side-lobes outside the area. The assumed edge loss of 2 dB could be reduced by slightly increasing the coverage area.

Path Loss

The space loss, 206.0 dB, is taken as an average mid-band loss over the coverage area and is also applicable for the worst rain-rate latitudes. Worst case space loss (e.g., the tip of Maine) is about 0.3 dB more but not in the high rain-rate areas. Mid-band loss is used both here and later for receiver antenna gain, because the frequency dependent terms perfectly compensate each other.

A rain fade of 2.5 dB (above normal atmospheric loss) is used for region 2 rain-climatic-zone M.[4] This figure is for a 1% outage during the worst month with a rain-rate of 63 mm/hr. A narrow band (region "N") along part of the Gulf of Mexico is about 2 dB worse and may require a 0.75 m receiver dish instead of the 0.6 m dish planned for use elsewhere.

The rain depolarization loss of 0.5 dB is conservative, and may actually be less at DBS frequencies.

Terminal G/T

The noise figure of 1.5 dB results from a survey of several companies which determined that the present state-of-the-art is about 0.7 dB for the new AlGaAs and InGaAs HEMT devices (High Electron Mobility Transistor). Using these devices, Sony claims a typical receiver noise figure of 1.6 dB for a new line of low noise converters. All of those surveyed agreed that 1.5 dB is a reasonable figure to assume in the near future for relatively inexpensive low noise amplifiers and converters at 12 GHz.

Sky and background temperatures are typical figures used at this frequency. However, rain temperature is more controversial. Spilker[5] claims that it could be as high as 100 degK, whereas Gagliardi[6] claims a figure of 10 to 50 degK. Using the most recent reference (Gagliardi) we have used a figure of 50 degK.

Receiver antenna gain is calculated for a small dish size of 0.6 m, using current, high-efficiency (75%) offset-fed terminal designs ready for production. A totally integrated antenna/down- converter is assumed. Therefore, no additional cabling and/or connector losses are incurred. A dish size of 0.75 m in worst case regions (e.g., rain-climatic-zone N) would increase the available margin by an additional 2 dB.

Signal-to-Noise Ratio

C/No (carrier power to noise spectral density ratio) is computed from EIRP, path loss, terminal G/T, and Boltzmann's constant. Eb/No (bit energy to noise spectral density ratio) is computed from C/No and the transmission bit rate.

Required Eb/No

Some video data compression algorithms can be quite fragile in noise. A single bit error in a sensitive part of the frame could cause a noticeable streak in the image if not promptly removed. To keep the rate of such events to less than once every 5 minutes, the bit error rate must be less than 10^-10. Error concealment can further reduce observable errors to less than once per hour. A suitable Reed-Solomon code (rate 7/8, 2048-bit block) requires a raw bit error rate of better than 0.15% (Berlekamp[7]) to attain this BER level.

For this BER, the figure of 7.5 dB for required Eb/No is based on a type of QPSK modulation[8] at 30 Mbps (total bit rate including error correction overhead) in a restricted bandwidth of 24 MHz. This assumes transmission pre-filtering, proper receiver equalization, and the TWT operating in saturation. While this figure is about 1 dB better than that computed by TRW[3] at this bit rate, the latter did not assume receiver equalization and did not optimize the transmitted waveform for this type of transmission.

TRW had originally also investigated transmission at a rate of 40 Mbps. It was found that this increased the required Eb/No by about 3 dB. This is considerably more than the power ratio indicates. The reason is that the selected transmission rate of 30 Mbps is only slightly higher than 1 bit/Hz where the bandwidth limitation is not a major factor.

40 Mbps transmission (for each of two channels for HDTV) would have pushed operation further into the bandwidth limited region where higher signal-to-noise ratio is required to overcome the restricted bandwidth. Fortunately, compression algorithms have improved enough that compatible digital HDTV is still considered to be practical using two 30 Mbps channels.

Margin

In the main coverage area with clear weather, a substantial margin of 7.7 dB is available. In the worst case edge-of- coverage with moderate rain, a margin of 1.4 dB still exists while using the same G/T terminal. This is a reasonable figure for margin and may actually increase in future years as noise figures further improve.

In a heavy rain rate area, a slightly larger 0.75 m dish would provide an additional 2 dB of margin and would further decrease rain outage. This would be a customer purchase option, analogous to buying a larger, more expensive television antenna.

Basic Digital Receiver

A basic block diagram of a generic Digital television receiver is shown in Figure A2. The down-converter selects the desired channel which is demodulated into a serial bit stream. Frame synchronization is extracted and error correction is implemented. Data for other services is recognized and either extracted or discarded at this point. Television decryption is implemented if the subscriber is authorized to receive the channel.

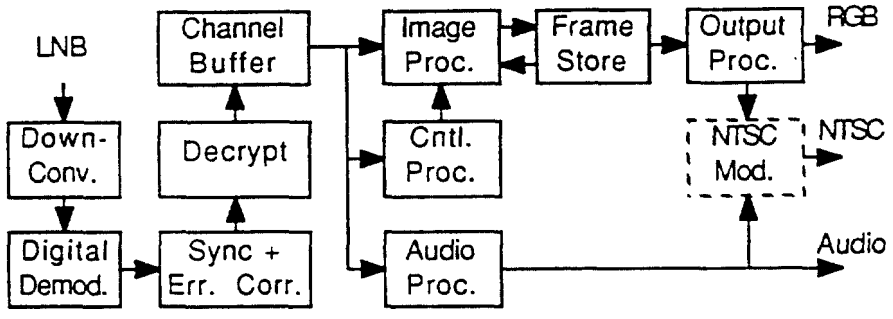

Figure A2 Digital DBS Receiver

A channel buffer allows the sometimes "bursty" image update data rate to be smoothed for transmission through the channel. The control processor interprets received command information and controls the immediate operations of the image processor. The latter processes the new received data along with the old data from the frame store to produce an updated image.

Luminance and chrominance information is processed and stored separately. Chrominance requires less resolution and is down-sampled by 2:1 vertically and 2:1 or 3:1 horizontally. Therefore, an output processor is necessary to reassemble the RGB data and D/A convert it to be displayed on a monitor. This can be done with either sequential or interlaced scan, depending on the capability of the monitor. For output to a standard (NTSC) television receiver, an NTSC modulator is required. These are very low-cost items and would be included.

Digital CD-quality audio is converted to a stereo analog signal which optionally can be modulated into the NTSC output.

Video Compression Techniques

A complete discussion of digital image processing techniques is beyond the scope of this attachment, as there is much information available and more is added daily. However, for television applications, some general trends are becoming evident.

Psycho-visual studies have shown that the human visual system can not observe high resolution in high motion or fast-changing scenes. Therefore, high resolution image transmission is not needed in those cases, particularly where the TV camera tends to blur these images. However, in stationary or slow-moving scenes, the eye is more sensitive to high-resolution information.

In stationary areas of the image it is possible and practical to start a scene change with low resolution and then refine it in succeeding frames. No further transmission is required if there are no more changes and the frame store will perfectly retain the image. Examples of means to accomplish this include subsampling techniques and transform techniques. In the latter case, the coefficients of the lowest frequency basis functions are transmitted first and the others follow in succeeding frames. If no changes occur over a long period, it is desirable to periodically transmit updates in order to erase the effects of any channel errors which may have occurred.

In slowly moving areas, motion compensation techniques may be used. Direct or interpolated high-resolution linear translation of the moving area can be made from the frame store instead of reverting to low resolution transmission. This requires very little information transmission (i.e., a translation vector and identification of the area block or boundaries). If predictive techniques are used, the translated image serves as the basis from which the (transmitted) prediction errors are determined.

High motion (or scene change) areas can use low-resolution, real-time transmission. In this case, the corresponding high resolution (but irrelevant) information in the frame store is simply discarded. Once motion stops, high resolution is gradually restored within the viewer perceptual response time.

Sequentially scanned images are transmitted at 30 frames per second. Although interlaced scan was necessary in the early decades of television, it causes substantial degradation as expressed in the "Kell" factor (used in computing lines of resolution). Interlace-to-sequential scan conversion can be accomplished at the transmission facility for video source material that does not use sequential scan. The receiver can then use the frame store to convert back to interlace scan for NTSC televisions and monitors.

New television monitors are already on the market which contain frame stores to convert the NTSC signal to sequential scan before presentation on the screen. This type of frame store (in the TV set) is not required with Digital DBS, since this function is inherently provided in the receiver image decompression. However, the availability in production of such capability is proven.

Within the presented general structure, many potential compression techniques are available. Some of these are:

- Subsampling
- Interpolation
- Predictive coding
- Transform coding
- Variable length coding
- Run-length coding
- Conditional replenishment
- Hybrid Predictive/Transform
- Non-uniform quantization
- Vector quantization

The exact compression algorithm for digital television transmission need not be specified at this time. A flexible architecture for an image decompression processor for home and school receivers could be designed to implement many combinations of the above techniques, although only one would be executed at a time. The decompression microcode could be down-loaded periodically from the satellite to the image processor. The control interpretation tables could similarly be down-loaded to the control processor. Such down-loading capability would add little (if any) cost to the receiver but would vastly increase its utility.

The down-loading requires (at most) a few Kbytes of error- detection-coded information. Transmission would occur during slack periods between image transmission peaks and could be done as frequently as once (or more) per second so that receiver channel switching could be done without noticeable delays. Detected errors would simply add a slight delay until the next transmission. This would be a very infrequent occurrence due to the extremely low bit error rate requirement to which this system is designed.

Initial service would be offered using a reasonably good compression algorithm. If an improved version of the compression/decompression algorithm, or a compatible version of HDTV or 3D is developed later, no change in the receiver hardware is required. The down-loading is totally transparent to the subscriber and could be done even on a program-by-program basis.

The main computational load is in the compression part of the algorithm. The receiver decompressor, of which there may be millions in operation in the field, operates in only one mode at a time and simply responds to transmitted commands. Efficient image compression may require a significant amount of processing hardware and software. However, in a broadcast system, only one compression system (plus a shared backup) is needed per channel.

Related Video Compression Systems

Several video compression systems have been used commercially in recent years. Some of them with their capabilities and limitations are discussed here.

The original Picturephone was a monochrome device having approximately half the linear resolution of an NTSC TV picture. While direct PCM coding would have required at least a 15 or 20 Mbps transmission bit rate, DPCM reduced the bit rate to 6.312 Mbps, and inter-frame coding further reduced it to below 1.544 Mbps.

The total compression ratio was approximately 10:1 and could have been higher if the codec had been located at the source instead of at a toll connecting point after impairments from analog transmission and switching. The codec was optimized for a head-and-shoulders view with a stationary background. Low- resolution modes were available in case of substantial subject motion or camera panning.

Compression Labs Inc., later developed a full NTSC system for video conferencing which operated at the T1 rate (1.544 Mbps). This was a compression ratio of about 60:1, but was also optimized for a view of nearly still subjects and a stationary background.

Later, Widcom developed a codec which operated at the (PCM audio) rate of 56 Kbps. This was an incredible compression ratio of well over 1000:1. However, it was very limited in handling more than eye blinks, mouth movement and a limited amount of arm motion.

Bell Labs[9] has recently announced a video codec for composite NTSC for use at the T3 rate (44.7 Mbps) which is about a 2:1 compression ratio. It uses simple intra-frame and inter-field coding techniques. While this is not an impressive compression ratio, it was motivated by the prevalence of the T3 (DS-3) rate in the common-carrier digital hierarchy, and by the need for virtually perfect reproduction in that service. The application is for fiber optic transmission of network (and other) television as a common carrier. Thus, the ultra-high fidelity requirement.

Two compression schemes that have been proposed for HDTV are the TRW algorithm[10] and the NHK "MUSE" system[11]. The former is a totally digital system using a programmable processor. It

uses mainly predictive and subsampling conditional replenishment techniques to achieve an HDTV compression ratio of about 10:1. Unfortunately, the system has not yet been built. Therefore, subjective comparison data to ascertain any visual degradation is not available.

The MUSE system is a hybrid analog/digital transmission system which relies mainly on subsampling over a four frame interval. The received signal is immediately digitized and processed in a similar manner to that shown in Figure 1 (including the frame store). Although the concept of "digital compression ratio" is not directly applicable to the MUSE system, an analog bandwidth compression of about 4:1 is achieved without noticeable degradation.

The most startling development was revealed recently at the second Microsoft CD-ROM conference[12], where RCA demonstrated a 70-minute, full-motion CD-Video. Observers said that the video quality was comparable to that of a VCR playback.

RCA also has developed a two-IC chip set to do all the image reconstruction in real time. The algorithm was not discussed, but it was stated that a substantial amount of off-line computer processing was needed to implement the video compression before pressing the CD. Apparently, RCA has been quietly working on this for several years. The resulting compression ratio is about 60:1 for full-motion, VCR-quality video.

The RCA system (at 60:1 compression ratio) certainly lends credence to a digital satellite system. Although Digital television has somewhat higher quality requirements and operates in real-time, a compression ratio of only about 5:1 is required. This should not be a problem to economically achieve in the very near future.

A compression system that is closely matched to DBS needs is described by Murakami, et al[13]. Although it has many similarities with the TRW codec (proposed for HDTV), it has significant differences as well. The Murakami codec has been implemented in hardware and has been subjectively evaluated. It is presently in commercial service in Japan.

The Murakami codec uses a compromise adaptive predictor. This results in suboptimal reconstruction prediction, but has the advantage of not requiring predictor identification overhead. A frame store and block motion compensation are used. Variable length coding is used for transmitting quantized predictor errors.

At 30 Mbps (video, audio and error correction), subjective evaluations of the Murakami codec showed no significant impairment. No jerkiness, dirty window, or edge busyness effects, which sometimes plague simple digital codecs, were observable.

At 15 Mbps, some coding noise and loss of resolution was perceptible for fine texture with large coloring and slow motion. Otherwise, the subjective evaluation was the same as 30 Mbps.

Although a wide spectrum of video codecs has been examined in this attachment, it is clear that 30 Mbps compressed digital television is both possible and practical.for full television movie transmission In fact, 15 Mbps will be more than adequate for most educational movie programs. For classroom lecture purposes (with more limited motion needs) Rates in the 1 to 3 Mbps range should be more than sufficient.

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12 Perry, Tekla, "Sarnoff research center reveals its DVI secret," IEEE: THE INSTITUTE, May 1987.

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Appendix B - Analysis of Dynamic Multiplexing

The number of channels that can be contained in a fixed allocated block (e.g.,a transponder or an ACTS allocation) can be determined by statistical analysis. The applicable analytical techniques are determined by basic assumptions about the active services. The type of service, basic fixed rate requirement, rate of request for higher bit rate, average high-rate burst length, and response to request denial are all relevant parameters.

As an example, consider the system shown in Figure B1 with sub-channelization shown in Figure B2. In this case a 30 Mb/s resource is divided into 20 fixed 1 Mb/s sub-channels plus six 2 Mb/s demand channels. By dynamic multiplexing, 3 Mb/s video quality can be delivered with an average transmission rate of only 1.5 Mb/s per channel.

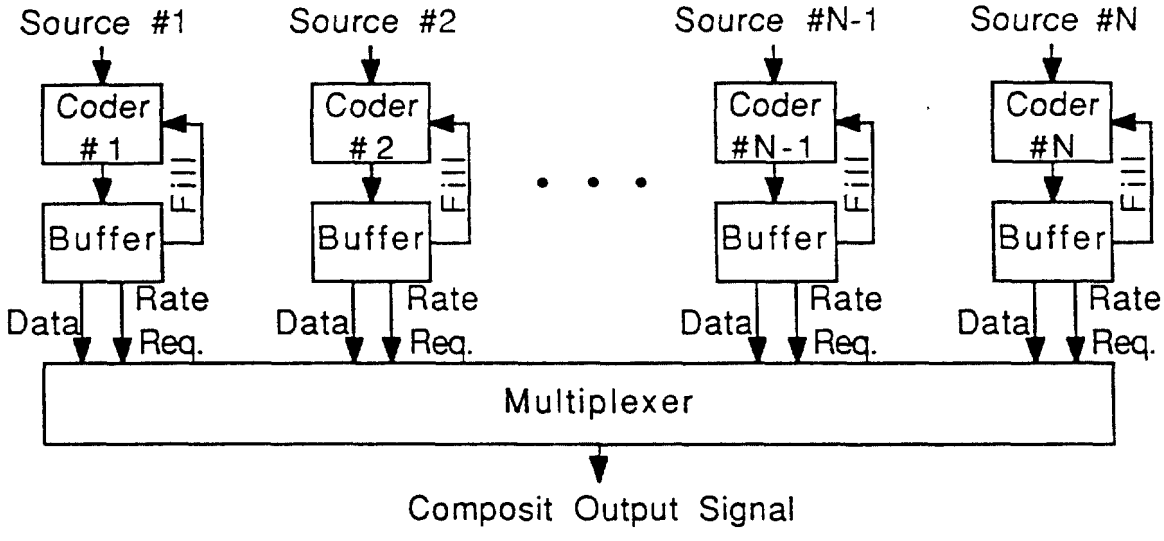

Figure B1 Example of Dynamic Multiplexer

For conferencing applications, there are video codecs available today which operate at rates as low as 64 Kb/s. However, the video quality is generally deemed to be inadequate at this rate. There is a substantial amount of interest in systems operating at 384 Kb/s. Using the same principles described here, 1 Mb/s video quality could be delivered with about 75 channels operating in a 30 Mb/s system.

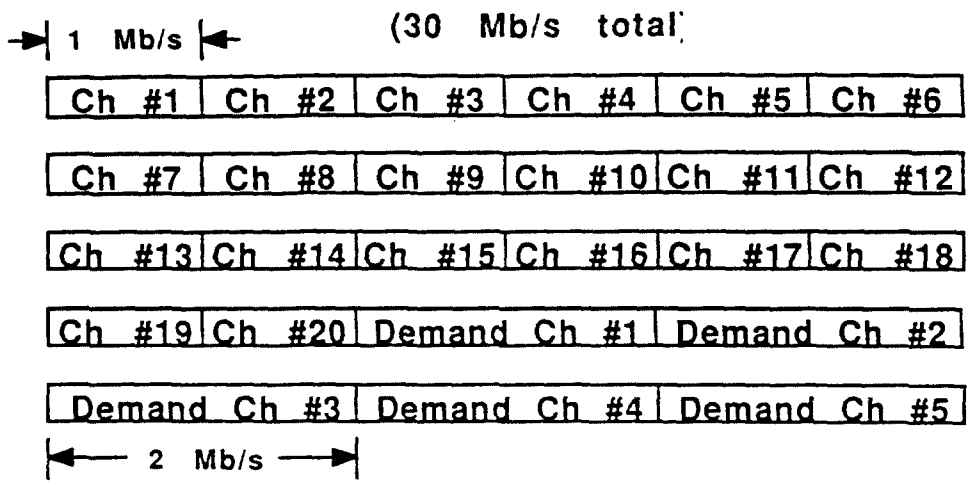

Figure B2 Example of Peak-Demand Channelization

This example system can be analyzed relatively easily for grade-of-service because of the identical types of sources and some not-very-restrictive assumptions about their behavior. The basic channel rates are assumed to be inviolate and occupy a fixed portion of the transmission allocation. The demand channels are also fixed-rate but are dynamically assignable to augment any channel on a first-come-first-serve basis. **Grade-Of-Service** (GOS) is defined as the probability that a significant motion event (i.e., request for a demand channel) is granted service by one of the demand channels.

If a request occurs when all demand channels are already occupied then the requesting channel may go into an "overload" mode where the source rate is choked down in order to prevent buffer overflow. This results in a possible visual degradation such as a loss of resolution in the motion area. Most of the following analysis will deal with the probability of motion degradation which is designated as "PMD" which is related to the defined GOS by:

$$GOS = 1 - PMD$$

The sources are statistically independent with random request times and the number of sources is large relative to the number of demand channels. Therefore, the requests can be assumed to have a Poisson probability distribution:

$$P_x = \frac{(RT)^x}{x!} e^{-RT} = \frac{\lambda^x}{x!} e^{-\lambda}$$

where R is the average (total) number of request per unit of time, the mean and variance are both $\lambda = RT$, and P_X is the probability of exactly x arrivals in time T. The probability density function of inter-request times is: