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Videoconferencing in the Energy Research Community

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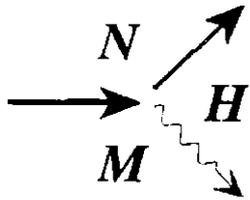
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August 1992

Developed as a contribution to the Remote Conferencing Working Group (RCWG) of the Energy Sciences Network Coordinating Committee (ESCC)

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1. INTRODUCTION

The energy research (ER) community is characterized by widely scattered groups of scientists working together on complex research topics. These collaborations have a need for frequent face-to-face meetings, yet travel is often prohibited because of time or expense. Advances in videoconferencing and network technologies have made possible a pilot project connecting several national laboratories in a videoconferencing network. This project has been highly successful with many other energy research sites wanting to participate. Because of this success, a production energy research videoconferencing network (ERVN) is being planned to utilize the existing videoconferencing systems and digital switched service. A central videoconferencing control center (VCC) will provide scheduling and management of the network. The new videoconferencing network will be flexible in both adding sites and upgrading the capabilities of existing sites.

2. ENABLING TECHNOLOGIES

Advances in long-haul network offerings, network termination equipment and video compression equipment have made the use of videoconferencing a viable alternative to travel and provide an advantage over simple audioconferencing.

2.1 Codecs and MCUs

The simplest form of video conference (shown in Figure 1) is called a point-to-point video conference. As the name implies, exactly two sites participate in the conference (one at each end of a long haul communications facility). The major component of the point-to-point video conference is the codec (COder/DECoder). This unit takes the analog signal from a video camera (the video signal) and the analog signal from a microphone (the audio signal) as inputs. It converts these input signals into a digital representation and uses signal compression techniques to reduce the bandwidth required for transmission. At the other end of the communications link a companion unit performs the required decompression, restores the information to an analog format, and uses it to drive the video monitor and conference speaker.

The more intricate form of video conference (shown in Figure 2) is when three or more sites participate in a single conference. This is called a multi-point video conference. The equipment at any site participating in the conference is the same as that used in a point-to-point conference. However, instead of connecting directly to each other, each site connects to a multi-point control unit (MCU). The MCU provides a mechanism to determine which picture is to be displayed on the conference monitor at each site. In practice, such units use some form of voice activated switching to determine what is displayed on the remote monitor at any site participating in the

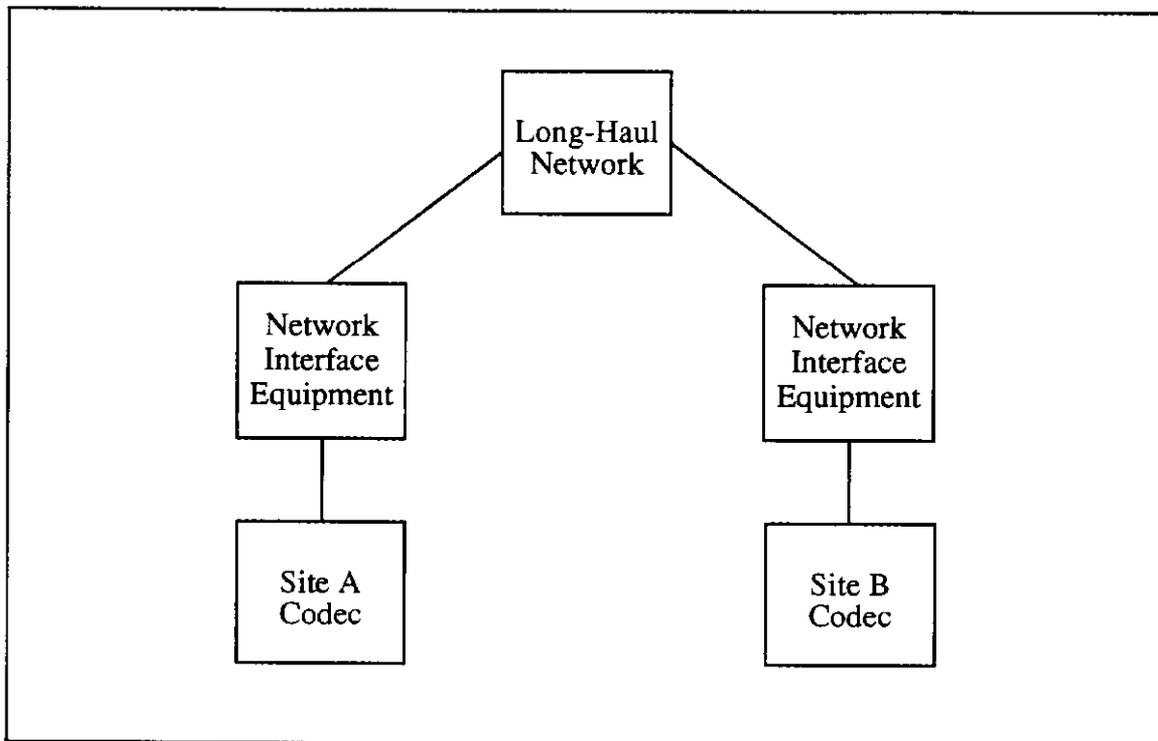


Figure 1. Point-to-Point Videoconference

conference. The algorithm used to determine what is displayed is quite simple: the picture displayed on the conference monitor is of that site where someone is speaking (or last spoke).

There are a growing number of video teleconferencing equipment manufacturers in the United States, including Compression Laboratories, PictureTel, and Video Telecom; there are also many foreign manufacturers (e.g. - Hitachi, Fujitsu). Due to the technical complexity involved in the signal compression process and the absence of early standards activity, each vendor has implemented a proprietary solution in its codecs. This means (at present) that all codecs involved in a video conference must be from the same manufacturer. If they aren't, things won't work. There is a great deal of activity on a CCITT standard, H.261, which is targeted at achieving codec interoperability. Vendors (including Video Telecom) have already demonstrated interoperability of their H.261-compliant equipment. H.261 has been approved by CCITT, but an audio compression standard to reduce audio bandwidth from 64 Kbps to 16 Kbps has not yet been approved. In addition, many codec manufacturers have additional features such as file transfer that are not supported by H.261.

Work is also being done on a standard for MCU interoperability, but until the standard is published and implemented, MCUs involved in a conference must be from the same manufacturer. This is viewed as a significant limitation when trying to provide for an open videoconferencing systems architecture. However, very few sites in the ERVN environment will have an MCU.

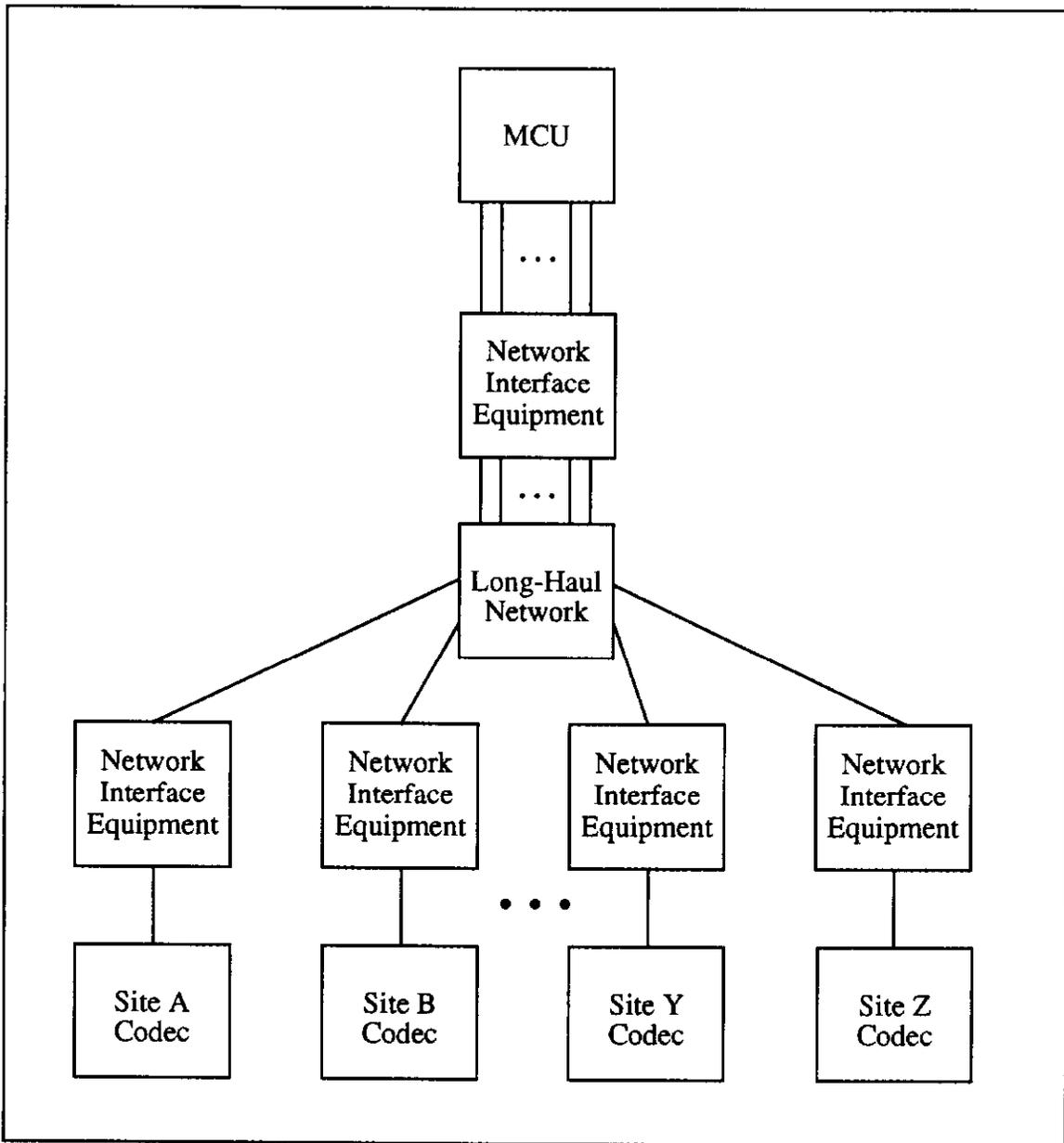


Figure 2. Multi-Point Videoconference

2.2 ISDN

Integrated Services Digital Network (ISDN) is a means of making the digital nature of most of the modern public switched telephone network (PSTN) available to end users. The bandwidth of an ISDN line is broken up into multiple channels, each channel being either type D or B. D channels run at either 16 Kbps or 64 Kbps and are used for call setup and control and for user packet communications. The B channel runs at 64 Kbps and is used strictly for user communications; ISDN uses no in-band signalling. In the United States ISDN service is delivered in two forms, basic rate interface (BRI) and primary rate interface (PRI). BRI service consists of two B channels and one

16 Kbps D channel. PRI typically consists of 23 B channels and one D channel, although there is a configuration where 24 B channels are made available from a single PRI and a D channel from another PRI is used for signalling (called non-facility associated signalling).

Many different types of service have been defined for the B channel, but few have been implemented. For example, Illinois Bell currently only offers 3 bearer services: 3.1 kHz audio, X.25 packet-switched data, and circuit-switched at 56 Kbps. For videoconferencing, we are only interested in one type of service, circuit-switched data.

ISDN is widely deployed in the networks of the US inter-exchange carriers (IXCs), such as AT&T, Sprint, MCI and Wiltel. Unfortunately, most US local-exchange carriers (LECs), such as Illinois Bell, General Telephone of Florida, and Pacific Bell, have not made ISDN widely available. Also, due to deficiencies in inter-switch communications, the US LECs currently offer only 56 Kbps B channel connections between their switches or between a single LEC switch and an IXCs network. Therefore, all US videoconferencing calls at bitrates higher than 112 Kbps must currently be done via bypass facilities. US sites that are part of National ISDN-1 trails, or international sites that have full 64 Kbps BRI access, do not have to worry about bypass for 128 Kbps calls.

In bypass, ISDN service from an IXC is delivered directly to the customer site via one or more T1 lines, each one carrying 23 B and one D channel. Although these T1 lines are often provided by a LEC, their use is controlled only by the IXC. Each site must have equipment to terminate this T1 and provide facilities to access individual B channels.

2.3 FTS-2000

The General Services Administration of the United States government has negotiated a contract with AT&T (60%) and Sprint (40%) to provide telecommunications services to all governmental and quasi-governmental agencies. The contract is known as FTS-2000. All U. S. Department of Energy FTS-2000 traffic is carried by the AT&T part of the network. The national labs do not widely use FTS-2000 at this time and are currently not required to do so. However, they may still order FTS-2000 services. Voice calls are transparently gatewayed from FTS-2000 to other national and international carriers. Unfortunately, data calls are not. Although the GSA is considering a change in the FTS-2000 contract to require interfaces to other carriers' data networks, currently all FTS-2000 digital data service (DDS) calls, such as ISDN, must stay on-network. Moreover, there is no interconnection between the AT&T and Sprint FTS-2000 ISDN networks. Likewise, the IXCs ISDN data networks are not interconnected, so, for example, an ISDN call cannot be placed from an AT&T Accunet ISDN user to a AT&T FTS-2000 ISDN user.

In addition to the data-only service provided by FTS-2000 ISDN, the FTS-2000 contract also includes compressed video transmission service (CVTS) which provides a "turn-key" solution using FTS-2000 data circuits and Compression Labs, Inc. (CLI) codecs and MCUs.

Several characteristics of the current CVTS offering make it unusable in the ERVN

- CVTS uses a CLI-proprietary coding scheme that does not interoperate with the current ERVN

equipment,

- CVTS has only a few MCUs currently and may not be able to handle the ERVN traffic,
- CVTS operates only at 384 Kbps.

2.4 Network Termination and Distribution Equipment

Once ISDN PRI is delivered to a ERVN site, it must be terminated, broken up into appropriate bandwidth, and distributed to the various codecs. Several types of equipment exist for this purpose. The traditional PRI termination and distribution mechanism is an ISDN-capable private branch exchange (PBX) or a central office (CO) switch used as a PBX. The PBX takes one or more PRI lines and breaks out pairs of B channels for distribution on-site via BRI lines. BRI lines use the unshielded, twisted pair (UTP) wiring that is standard for normal plain, old telephone service (POTS) so that this distribution mechanism integrates well into the sites POTS phone system.

A more sophisticated solution, and the only viable one for sites without an ISDN-capable PBX, is to use a PRI network access unit. Such a unit takes multiple PRI connections like the PBX, but offers many more options in distribution of the channels. In addition the unit also typically includes inverse multiplexing capabilities (to be discussed later) and complex bandwidth and network management. Although there are several manufacturers of PRI network access units, we will focus on the one recommended for the ERVN, the NetworkHub by Teleos.

The NetworkHub can take PRI lines and dynamically distribute the B channels over BRI, PRI, or T1 lines. It can also aggregate the bandwidth of multiple B channels (N×64, or N×56 Kbps) and make this bandwidth available as (1) V.35 with RS-366 dialing, (2) RS-449 with RS-366 dialing, or (3) BRI with D-channel dialing. Finally, it can route individual B channels to other PRI trunks or reroute bandwidth from a V.35 link to another V.35 link. The NetworkHub comes in several configurations with list prices from \$16,000 for a unit with 1 PRI port and 2 V.35/RS-366 interfaces to \$80,000 for a unit that will support several codecs and an MCU or two. Teleos has packaged the NetworkHub in a dizzying array of configurations, some called VideoHub, some called RemoteHub and some called only NetworkHub, but the product is the same except for switching bitrate and number of slots. The interface cards are interchangeable between all products.

One or more PRIs are delivered from the IXC to the site and terminated by a channel service unit (CSU) for each PRI. The CSU provides line isolation and loopback facilities and the conversion from a U (two-wire) to a V.35 interface. Each PRI is then fed into the NetworkHub for switching and distribution. Sites that currently have ESnet connections may be able to take advantage of ESnet bandwidth allocated for videoconferencing by using a T1 splitter to derive two 64 Kbps channels (DS0s) and provide them to the NetworkHub via a V.35 interface. The rest of the bandwidth would then be provided to the ESnet router connecting the site's data network to ESnet. This use of ESnet bandwidth is intended for use only during transition to switched service.

Teleos sells options for the NetworkHub that allow for both local and remote control through a dial-up interface. To be useful in the ERVN, the NetworkHub must be ordered with this option. Otherwise many configuration changes would require help from Teleos. Local and remote control

is provided by a graphical interface on a PC running specialized software, MS DOS and Microsoft Windows. Figure 3 shows a sample configuration. The dotted lines represent equipment that may not be required.

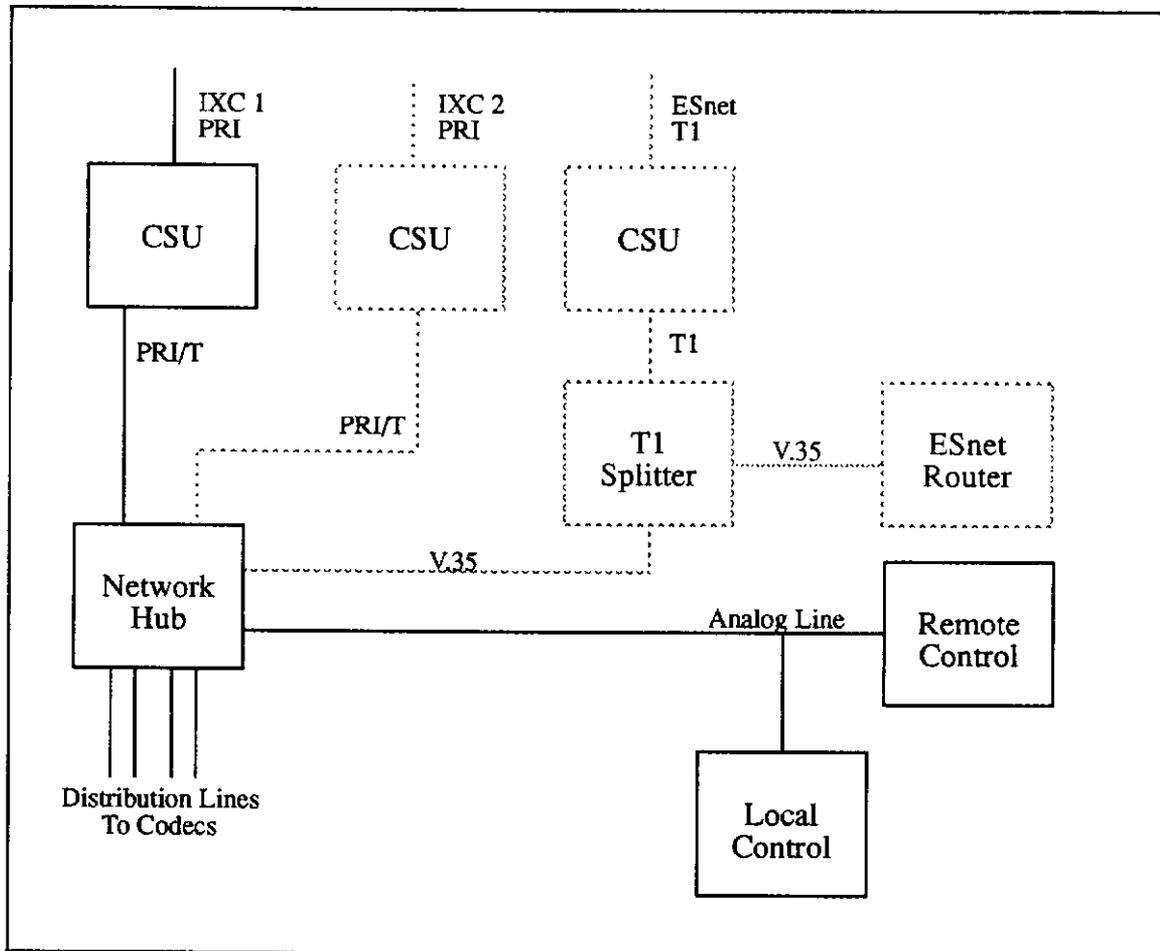


Figure 3. NetworkHub Configuration

Teleos also makes a product called the HubExtender that extends the V.35/RS-366 interface of the NetworkHub up to 3000 wire feet, via unshielded, twisted pair (UTP). The product is unique in that it will automatically adjust to different bitrates, from 112 Kbps up to 1.44 Mbps. The HubExtender is recommended for use when V.35/RS-366 distribution is used and the codec is more than 100 wire feet from the NetworkHub.

2.5 Inverse Multiplexers

Since video calls need more than 64 Kbps of bandwidth, multiple B channels must be aggregated together to provide the needed capacity. There are several points in the videoconferencing network where this can be done: in the T1 termination equipment, in the equipment connecting the codec to the network, in a stand-alone box between the network interface and the codec, or in the codec itself. Unfortunately, there is no widely-accepted standard for inverse multiplexing, so units

on each end of the link must be from the same manufacturer or have proven interoperability. A standards group called BONDING (bandwidth-on-demand inter-networking group) will be introducing a set of standards sometime late in 1992, but even then it may be some time before the marketplace decides which mode of the BONDING mode is most desirable and all manufacturers implement it.

3. VIDEOCONFERENCING PILOT PROJECT

The Energy Sciences research community has made extensive use of electronic communication in the form of computer networking for over a decade. Due to the growing size and dispersion of scientific collaborations, videoconferencing is becoming a necessity in the conduct and management of large projects and programs. A number of economic factors are driving this change:

- fundamental advances in visual telecommunications technology
- rapid reduction in the cost of video teleconferencing equipment
- dramatic reduction in the cost of switched digital network services
- increased travel costs and awareness of productivity loss

3.1 Initial Pilot Project

In early 1990, a video teleconferencing pilot project was established between three ESnet backbone sites, Lawrence Berkeley Laboratory (LBL), Fermi National Accelerator Laboratory (FNAL), and the Superconducting Super Collider Laboratory (SSCL); see Figure 4. Each of the sites purchased a videoconferencing system (essentially a codec, cameras, television monitors, speakers and microphones); in addition, a multipoint control unit (MCU) was purchased so that all three sites could participate in a conference simultaneously. The data circuits needed for the pilot project were provided by allocating bandwidth from existing ESnet T-1 communications facilities. The pilot project produced a fuller understanding for videoconferencing requirements, and of associated technical and management issues; it also demonstrated that videoconferencing can often be a viable alternative to travel.

A study of video teleconferencing equipment from several vendors including Compression Labs, PictureTel, Video Telecom Corporation, etc. was done. Video Telecom Corporation (VTC) video teleconferencing systems were selected for several reasons:

- VTC had the only fully functional multi-point control unit (MCU) at the time. Multi-point videoconferencing was considered essential for collaborative applications.
- VTC systems were software driven (and ran on an IBM PC clone) allowing for ease of equipment upgrade which provides better performance and features
- VTC was committed to provide an equipment upgrade path to the H.261 codec interoperability standard
- The cost of VTC systems were substantially less than those of comparable systems in use at the time

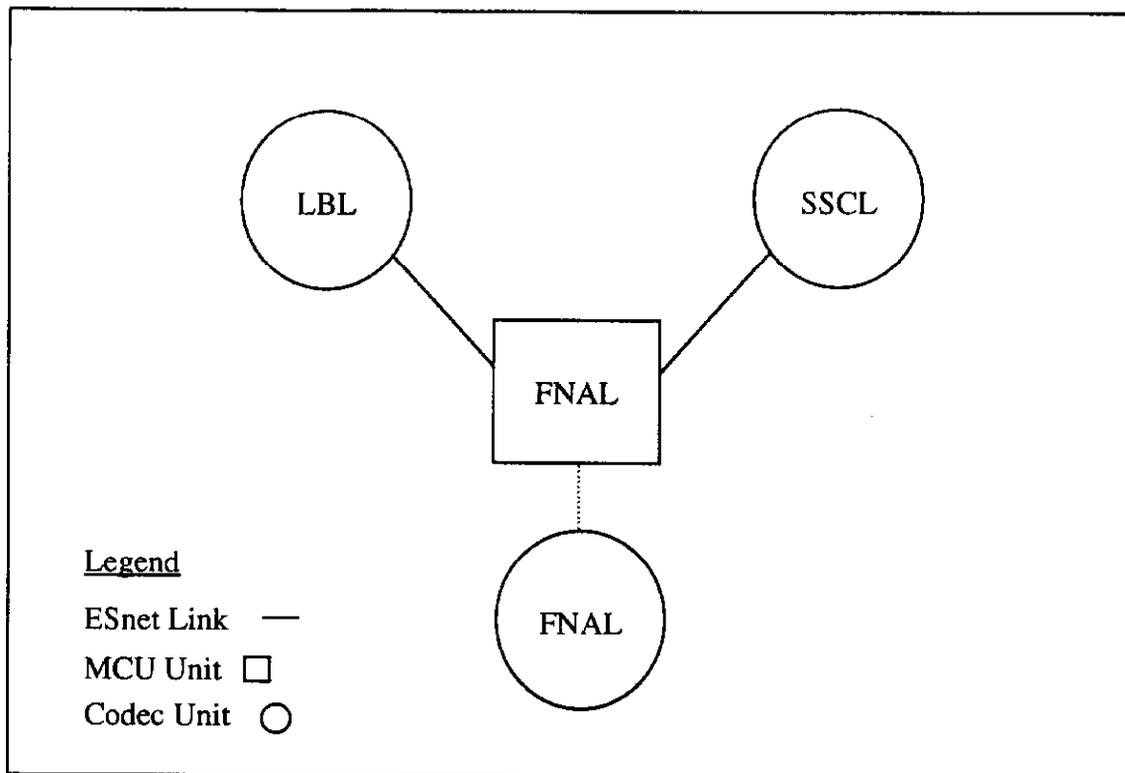


Figure 4. Initial Pilot Project Architecture

3.2 Current Pilot Project

The videoconferencing network currently in operation (see Figure 5) is an extension of the existing pilot project. The pilot has been expanded by installing a second MCU at the SSCL. Bandwidth was derived from selected ESnet T1s to connect three new sites, Argonne National Laboratory (ANL), Harvard University, and Oak Ridge National Laboratory (ORNL). In addition, videoconferencing to the University of Michigan (Ann Arbor, Michigan, USA), KEK (Tsukuba, Japan) and to INFN (Pisa, Italy) is provided through a dial-up ISDN interface at the Superconducting Super Collider Laboratory. The network has the current characteristics:

- All codecs and MCUs used are from Video Telecom Corporation.
- All codecs are the CS-350 model.
- Dedicated bandwidth is provided by carving one or more 128 Kbps channels out of selected ESnet T1 links.
- Dial-up bandwidth is provided by AT&T Accunet switched digital service.
- All videoconferences are at 128 Kbps.
- All connections between sites are made through MCUs.
- Bandwidth allocation and setup is done through T1 splitters.
- Scheduling of conferences, MCUs, and links is done in an ad-hoc manner.

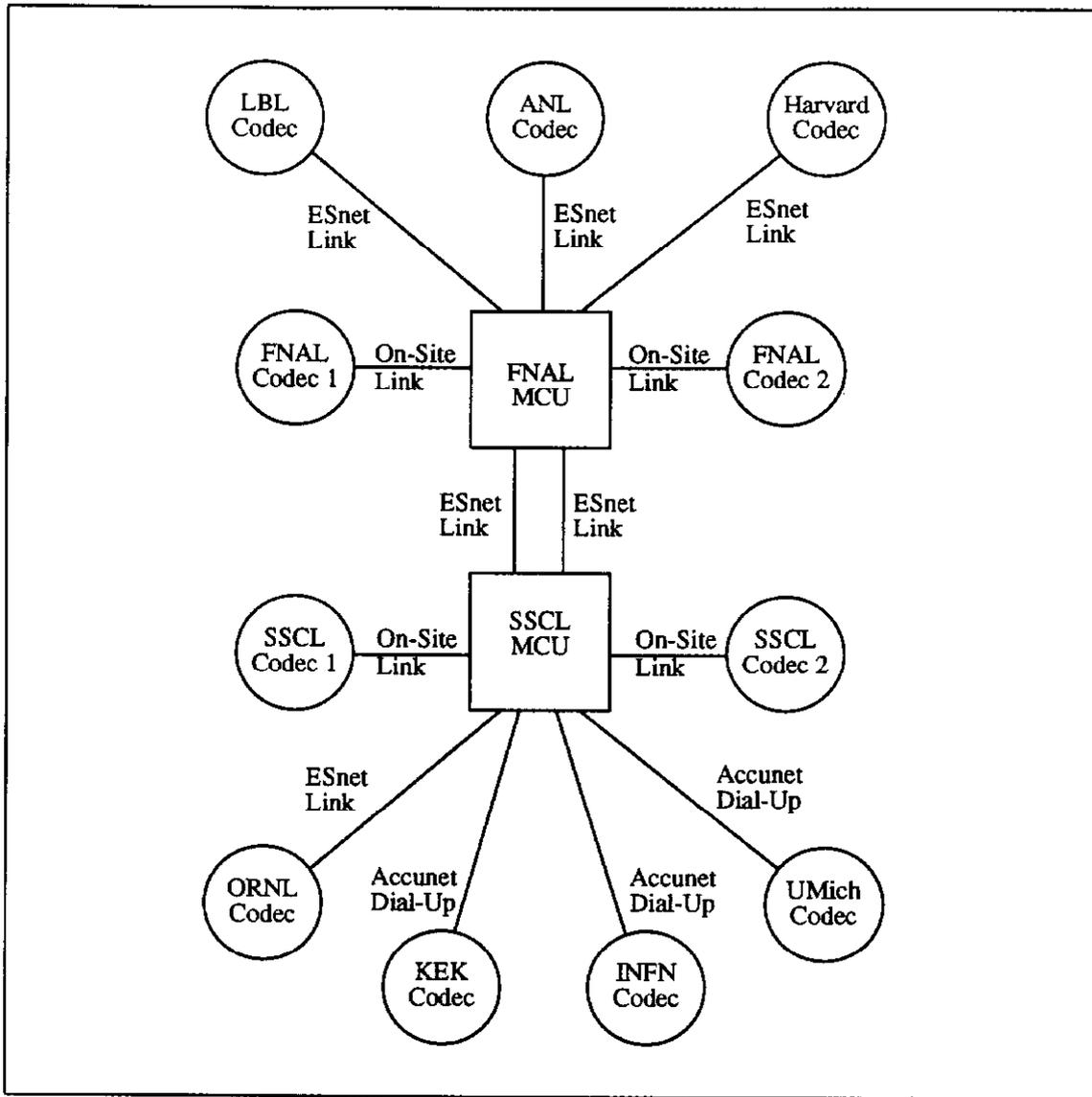


Figure 5. Current Pilot Project Architecture

3.3 Short Term Pilot Project Expansion

The demand for videoconferencing capability continues to grow and plans are being formulated to transition the video pilot project to a viable production service. The videoconferencing network will expand by five sites and add an MCU later in 1992 (see Figure 6). The new sites are the University of California at Irvine (UCI), the California Institute of Technology (CIT), the Massachusetts Institute of Technology (MIT), Brookhaven National Laboratory (BNL), and the Continuous Electron Beam Accelerator Facility (CEBAF). Tentative plans are for a third MCU to be located at MIT. Connections to CIT and Harvard may be made by using bandwidth from ESnet or through dial-up access to the SSCL.

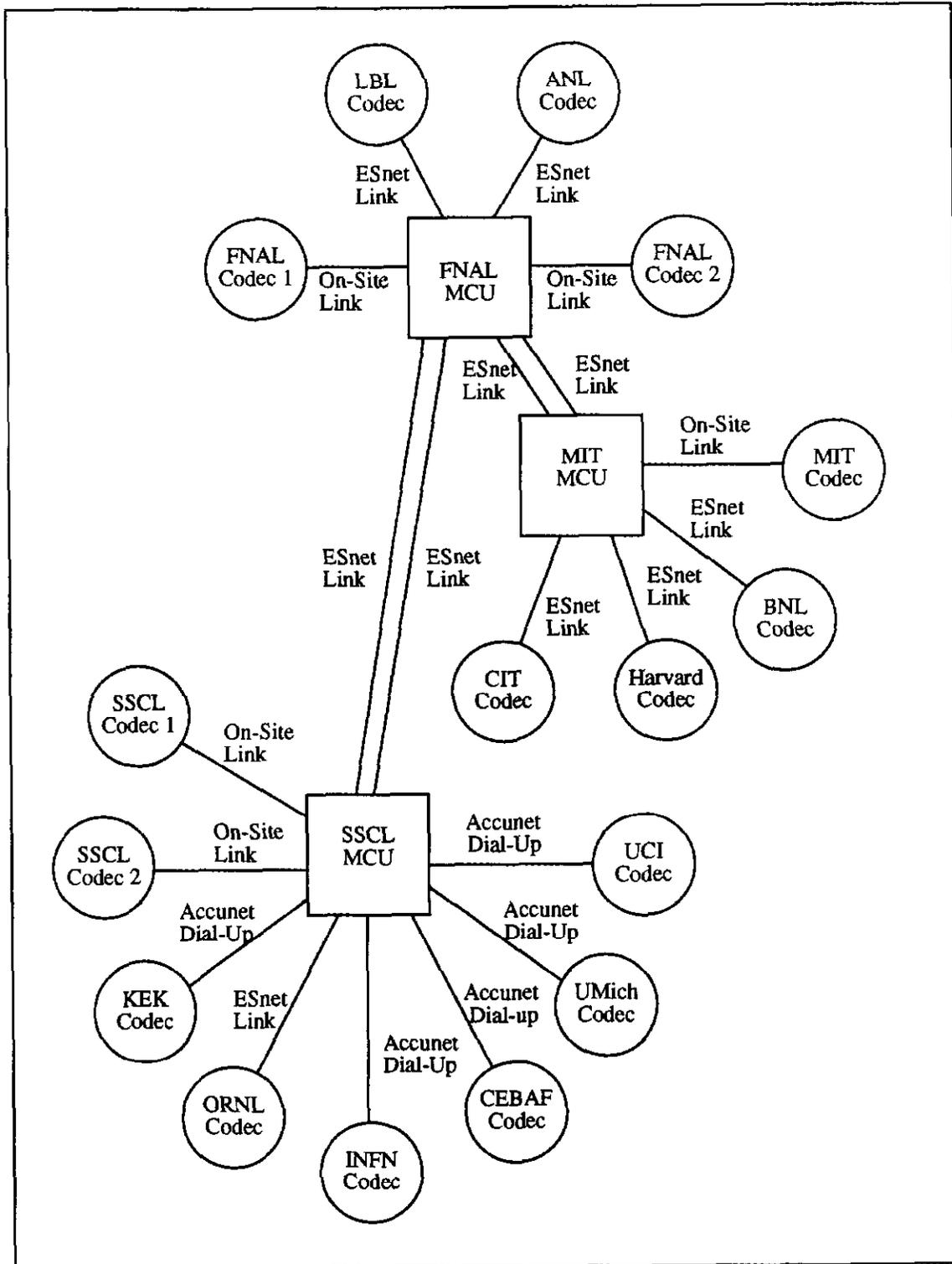


Figure 6. Future Pilot Project Architecture

3.4 Pilot Project Findings

The following information was learned as a result of the videoconferencing pilot project.

- Multi-point videoconferencing is essential to support large experimental collaborations.
- To minimize the costs associated with communications, the lowest acceptable bitrate was selected for operation. A bitrate of 128 Kbps was chosen as the universal bitrate of operation because compatibility with ISDN and the H.261 standard will be important in the long term.
- Scheduling and operation of the videoconferencing network require substantial effort as the number of sites increase.
- Using multiplexed channels out of point-to-point ESnet communications circuits without bandwidth management is not appropriate; it creates poor utilization of MCU ports and ESnet bandwidth and poses difficulties for sites not on the ESnet backbone that wish to use videoconferencing.

4. PRODUCTION SERVICE ARCHITECTURE

There has been a continued and rapid growth in the use of videoconferencing in the energy research community. This is evident both in the number of new sites acquiring videoconferencing equipment, and in the increased usage of the equipment at sites having videoconferencing systems. Much has been learned about the deployment and use of the videoconferencing technology during the pilot project. It is now necessary to establish a production quality videoconferencing service for energy research.

Some important goals of a production quality videoconferencing service are:

- Protect existing capital investment in video teleconferencing equipment;
- Architecture must be flexible enough to upgrade gracefully as video teleconferencing technology matures;
- Provide a centralized management point for setup/teardown of video conference "calls", and for conference reservation;
- Make appropriate use of dedicated and switched communications services where economically appropriate;
- Provide capability for participation in videoconferencing to DOE and NSF supported research sites not connected to the ESnet backbone;
- Provide capability for participation in videoconferencing for DOE administration sites;
- Migrate to the H.261 codec interoperability standard when appropriate;
- Support interoperability of Energy Sciences video teleconferencing equipment with systems from other vendors, important because it permits conferences with outside vendors and universities that may have chosen a different vendor;
- Consider changes in video teleconferencing environment as carriers deploy new services (e.g. Frame Relay, Asynchronous Transfer Mode, Switched Multimegabit Data Service);
- Provide private (but not secure) point-to-point video connections.

In order to meet these goals, the architecture of the ERVN will have to change dramatically. Instead of the current fixed topology, with codecs at a site directly connected to a specific MCU port at a specific site, the architecture will have to be more flexible, allowing a codec at a site to dynamically connect to a codec or MCU at a number of sites, thus forming a videoconferencing virtual network “cloud” (see Figure 7). Likewise, instead of the current scheme where all conferences are fixed at 128 Kbps, videoconference calls may be placed at a number of bitrates.

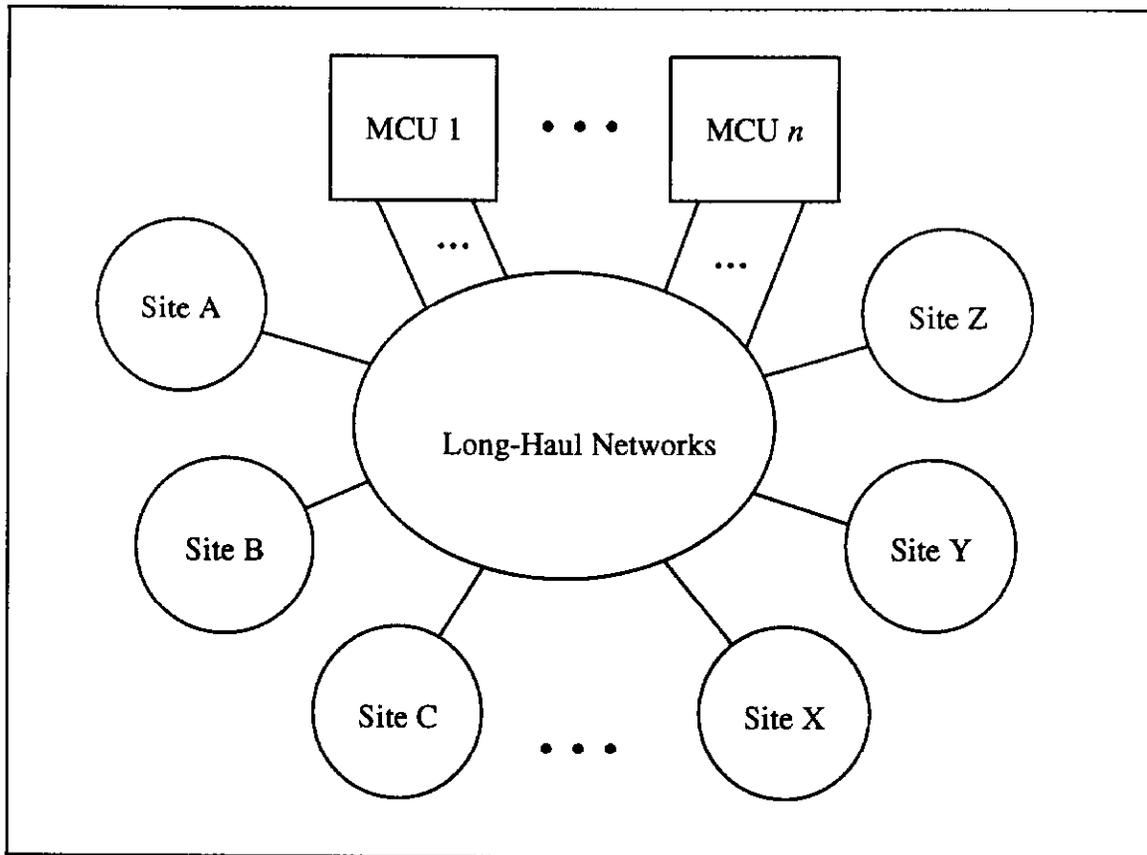


Figure 7. ERVN “Cloud”

The core users of the ERVN are expected to have videoconferences with the following characteristics:

- Codecs will be CS-350s using the Video Telecom CS-350 coding scheme;
- Codec bitrates will be either 128 or 192 Kbps;
- Long-Haul communications facilities will be provided by AT&T FTS-2000 switched digital service;
- Access to the long-haul network will be via PRI through a Teleos NetworkHub;
- Inverse multiplexing will be done in the NetworkHub.

As soon as feasible, the CS-350s will be replaced with MediaMax's and the coding scheme will change from CS-350 to Blue Chip. Sites that do not conform to these requirements will still have full access to the ERVN, but must go through a gateway to communicate with core sites.

4.1 Videoconferencing Control Center

As the number of sites within the Energy Sciences research community having videoconferencing equipment grows, the complexity of managing the video network also grows. When properly configured, the hardware outlined in the previous section permits an energy research or DOE administration site to join what can be viewed as a videoconferencing network "cloud," with AT&T FTS-2000 ISDN switched data service as its primary data transport. In addition, it has a very sophisticated set of customer premises equipment at each site. Management of the equipment for all sites connected to the "cloud" is a complex problem.

Central to the ERVN is the videoconferencing control center (VCC). The VCC will have the responsibility of

- scheduling videoconferences,
- bridging multi-party conferences together,
- setting up and tearing down videoconferences,
- configuring the dedicated bandwidth from ESnet during the transition to switched service,
- dynamically choosing network facilities and carriers,
- gatewaying between different carrier networks,
- maintaining and repairing network facilities,
- providing information on what equipment to order and how to configure it,
- negotiating with vendors on upgrade and interoperability.

4.2 Gatewaying

For two sites to be able to hold a directly-connected videoconference, there must be compatibility in several different areas:

1. they must be using the same video and audio encoding scheme (i.e. H.261, Video Telecom CS-350 proprietary, Video Telecom Blue Chip, CLI proprietary),
2. they must be running at the same bitrate (i.e. 112, 128, 192 Kbps).
3. they must be on the same network (e.g. AT&T FTS-2000, Sprint FTS-2000, AT&T, Sprint, WilTel, MCI, ESnet),
4. they must be able to establish a connection,
5. they must use the same inverse multiplexing scheme (i.e. Video Telecom proprietary, Teleos proprietary, BONDING mode 1-4).

Core sites should be compatible in all areas, but other sites may not be. The VCC can help alleviate some of these stringent requirements by providing gatewaying services. Gateways between codecs using different encoding schemes or different bitrates are difficult to provide, but several

commercial videoconferencing network providers offer such a service (e.g. the Sprint Meeting Channel). The VCC may contract with an outside network provider to supply a videoconferencing gateway service.

The requirement of being on the same network is easier to soften. By having connections to several networks, the VCC can bridge calls from one network to another. Adding to this, the ability to place data calls relieves the requirement that the sites be able to establish a connection. The VCC may set up the connection for both sites. In fact, this may be preferable to provide centralized billing to the VCC. Finally, the VCC should have equipment to handle several different inverse multiplexing schemes and be able to dynamically switch this equipment among IXC networks

Table 1 lists the gateways that the VCC should provide to allow non-core ERVN users to conference with one or more core ERVN sites. Note that Blue Chip coding scheme gateways will replace the CS-350 coding scheme gateways when the transition from CS-350 to MediaMax occurs. The pilot project gateway will be provided only during transition to a fully switched videoconferencing network. Gateways will provide only audio and video capabilities. To fully utilize the resources of the videoconferencing systems (graphics transfer, shared drawings, etc.) like systems are necessary.

Service	Coding Scheme	Transport	Bitrate (Kbps)	Inverse Mux	Note
Pilot Project	CS-350	ESnet	128	None	only supported during transition
Accunet ISDN	CS-350, H.261	AT&T Accunet	112, 128, 192	Codec, Teleos	Primarily for International Gateway
FTS-2000 ISDN	CS-350	AT&T FTS-2000	112, 128	Codec	Primarily for sites without a NetworkHub
FTS-2000 CVTS	CLI	AT&T FTS-2000 T1	384	None	Primarily for DOE Offices
Sprint Meeting Channel	CLI, PictureTel, H.261	Sprint T1	112, 128, 192, 384	None	Primarily for outside vendors

Table 1: VCC-Supplied Gateways

Typically, conferences will be at 128 or 192 Kbps via FTS-2000. The 112 Kbps option is for those low-usage sites that do not want to get PRI service and can get only 56 Kbps BRI or Switched 56 service from their LEC. The 384 Kbps option is for sites that want higher quality (and higher cost)

video, although the improvement over 192 Kbps is not great. As sites move to access schemes allowing 384 Kbps calls and as bitrate costs are reduced, the ERVN may move to mostly high-bandwidth video calls. Core ERVN sites will use their NetworkHub for inverse multiplexing, except in a few special cases. All calls at bitrates in excess of 128 Kbps will use the Teleos scheme, since the Video Telecom codec cannot inverse multiplex for bitrates over 128 Kbps.

The VCC should be sensitive to the user and potential user communities and add more gateways as needed. Note that many of these gateways could be done at individual sites. However, unless a large amount of traffic warrants a local gateway, it is recommended that the VCC be used to connect to different networks or to sites using different inverse multiplexing schemes.

4.3 Scheduling and Control

The issue of videoconference scheduling is very difficult to address. The stumbling block is that one of the entities being controlled is a conference room and video conference system that has been purchased by the site, not the VCC. Ideally, the VCC should provide scheduling and control of all network facilities, codecs, and videoconference rooms. However, this may not be practical, especially for those sites that may also be part of another videoconferencing network. Therefore, each site should select a site videoconferencing coordinator who will, through cooperation with the VCC, provide room scheduling and help with local equipment issues.

Although the scheduling and equipment set-up functions of the VCC will initially be manual, eventually a commercially-available videoconference management package will be purchased to facilitate control of the resources in the ERVN. This software will be used to schedule the use of all of the video conference facilities connected via the “cloud” described previously. If a given site has periods of time when it is being used in private conferences, then the site will use the centralized scheduling mechanism to declare itself unavailable. The same would be true if the conference room at a given site were being used for some other activity, or the video equipment wasn't operational. The VCC management facility would examine the consequences of a given site being unreachable, and notify other participants in a scheduled conference of the problem. A decision could then be made to hold the conference anyway, or to reschedule it at a later date. Access to the VCC scheduling system will be provided via ESnet and via dial-up.

Because of digital switched service pricing, it may be advantageous for the VCC to place all calls to set up a videoconference. In fact, even point-to-point videoconferences might be cheaper if two calls are placed from the VCC and bridged together rather than a single call from one site to the other. These economies of scale, and the coordination required to take advantage of them, is one of the main reasons for establishing the VCC.

5. SITE DISTRIBUTION ARCHITECTURES

There are many options for connecting to the VCC and distributing the bandwidth on site. This section presents several architectures that are appropriate in different scenarios. Almost infinite

variations on these configurations exist, but before sites implement a modified configuration the VCC must be involved. All dashed lines in the figures are equipment that may not be needed.

Most of the configurations involve the NetworkHub. It is highly recommended that US sites that will be frequent users of the ERVN obtain a NetworkHub and a PRI connection to the AT&T FTS-2000 ISDN network. This will ensure compatibility with other sites and allow the VCC to provide a better level of management and support. The other configurations shown are for international sites and US sites that will be infrequent users of the ERVN or that have special requirements. Table 2 shows the options discussed in this section and the relative costs of each.

Option	Bitrate (Kbps)	Inverse Mux Point	Equipment	Relative Cost	Must Go Through Gateway
5.1, NetworkHub-Based Distribution via a Single V.35/RS-366 Line (RECOMMENDED)	112-384	Teleos	Codec, VideoHub, HubExtenders	Med-High	No
5.2, NetworkHub-Based Distribution via Two V.35/RS-366 Lines	112-384	Teleos or Codec	Codec, VideoHub, HubExtenders	High	No
5.3, NetworkHub-Based Distribution via BRI	112-128	Codec	Codec, VideoHub, Terminal Adapter	Med	Yes
5.4, NetworkHub-Based Distribution via PRI	112-384	Teleos or Codec	Codec, VideoHub, RemoteHub	Very High	No
5.5, PBX-Based Distribution via BRI	112-128	Codec	Codec, PBX, Terminal Adapter	Med-Low	Yes
5.6, LEC-Based Distribution via BRI	112-128	Codec	Codec, Terminal Adapter	Low	Yes

Table 2: Site Distribution Options

5.1 NetworkHub-Based Distribution via a Single V.35/RS-366 Line

Figure 8 show the configuration recommended for most ERVN sites. In this configuration, multiple B channels from the IXC PRI are inverse multiplexed by the NetworkHub and then routed to the appropriate codec via a V.35/RS366 line as an aggregate bandwidth. A set of HubExtenders must be used to extended the line for each codec that is not co-located with the NetworkHub. This configuration meets all the requirements of being a core ERVN site at the lowest cost.

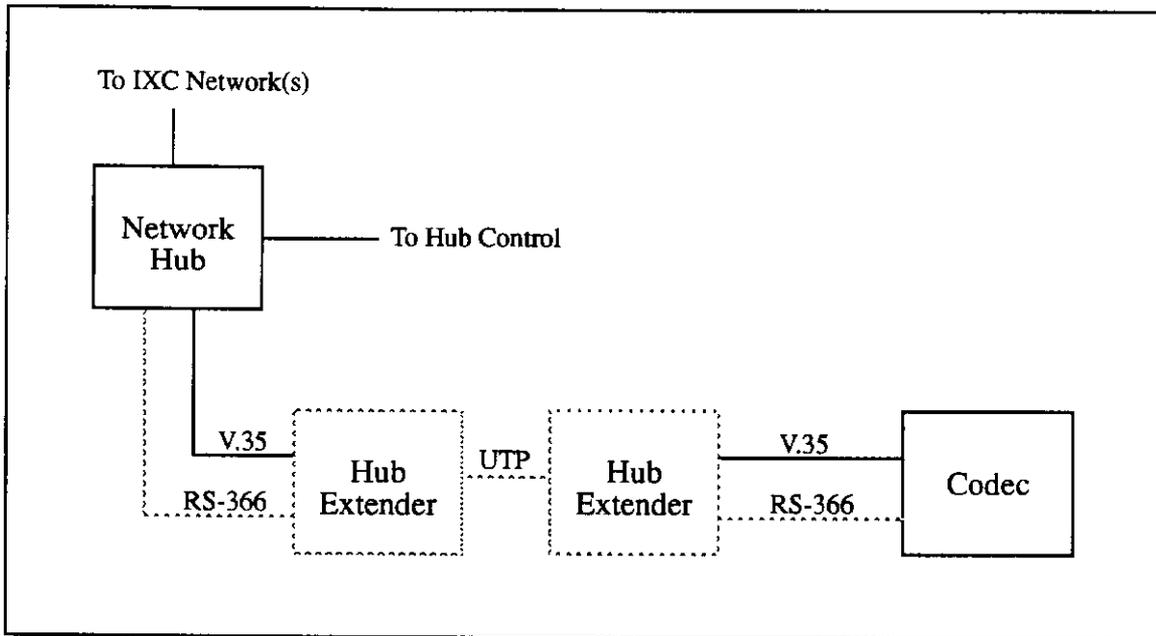


Figure 8. NetworkHub-Based Distribution via a Single V.35/RS-366 Line

5.2 NetworkHub-Based Distribution via Two V.35/RS-366 Lines

Figure 9 gives a more versatile configuration. For 112 and 128 Kbps calls via two B channels, the NetworkHub will route one B channel over each V.35/RS-366 interface as a 64 Kbps synchronous data stream. Inverse multiplexing will be done in the codec. The codec may make calls using the RS-366 dialing scheme which the NetworkHub will translate to ISDN signaling for the IXC PRI. Alternately, 112 and 128 Kbps videoconferences can be placed using the NetworkHub to inverse multiplex the two B channels together and providing the 112 or 128 Kbps of bandwidth via a single V.35/RS-366 interface; the other V.35/RS-366 interface will not be used. The codec may still place calls using the RS-366 interface, but will not do the inverse multiplexing. For 192 Kbps calls, the NetworkHub will inverse multiplex three B channels together and send the signal out over one V.35/RS-366 interface running at 192 Kbps. Again, only one interface will be used, but dialing is still possible.

Advantage over 5.1:

- Can also have point-to-point videoconferences with sites using Video Telecom multiplexing scheme.

Disadvantage over 5.1:

- May have to buy 2 sets of HubExtenders for each codec,
- Uses two V.35/RS-366 ports on the NetworkHub for each codec,
- Requires codec with inverse multiplexing capability.

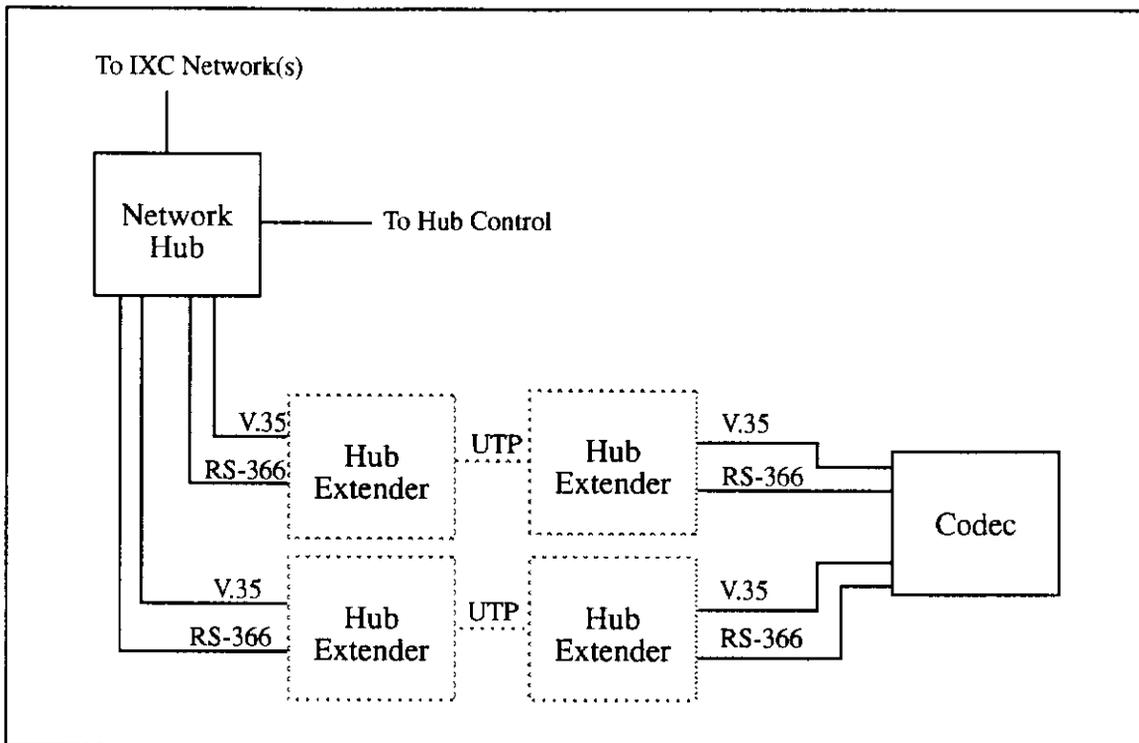


Figure 9. NetworkHub-Based Distribution via Two V.35/RS-366 Lines

5.3 NetworkHub-Based Distribution via BRI

The least costly NetworkHub configuration is presented in Figure 10, distribution via BRI. In this configuration, the NetworkHub merely acts as a way to redistribute the B channels from the PRI. The pair of calls representing a videoconferencing call would be routed out over a single BRI (2B+D) line. An ISDN terminal adapter would then make the 64 Kbps from each B channel available as a V.35 connection. The terminal adapter would also convert the RS-366 signaling for each V.35 line into ISDN D-channel messages to the NetworkHub. Note that the BRI lines from the NetworkHub are separate from those that may be available at a site from a CO or PBX and are limited to a distance of 3000 wire feet.

Advantages over 5.1:

- Uses only one BRI port on NetworkHub for each codec,
- Does not require HubExtenders,
- Can have point-to-point videoconferences with sites using Video Telecom multiplexing scheme.

Disadvantages over 5.1:

- Requires a terminal adapter for each codec,
- Must go through gateway to communicate with core ERVN sites,

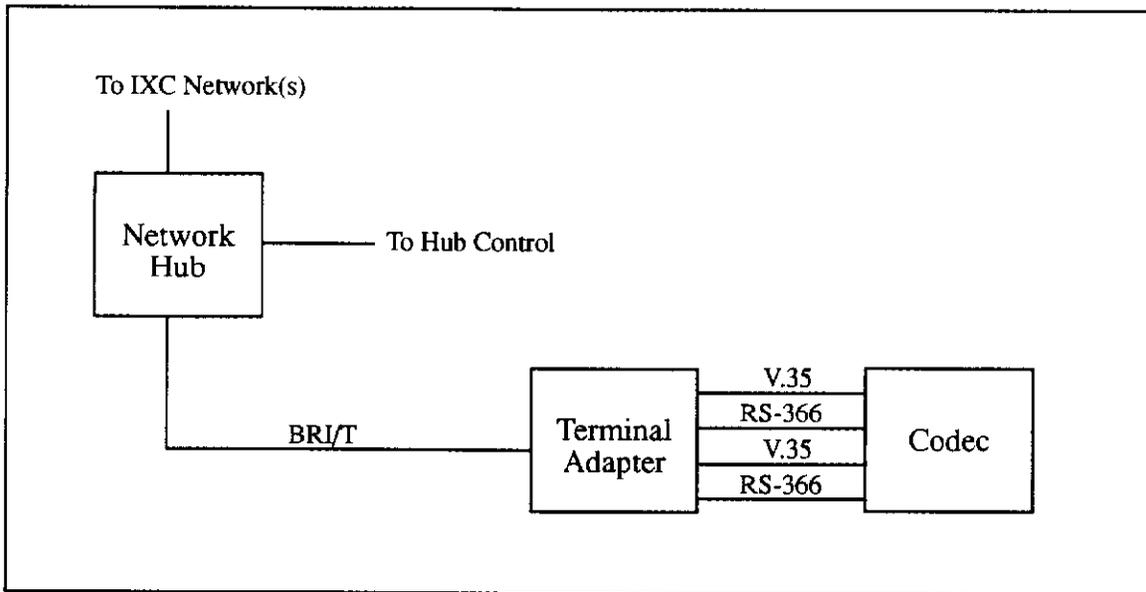


Figure 10. NetworkHub-Based Distribution via BRI

- Cannot be used for videoconferences at bitrates greater than 128 Kbps,
- Requires codec with inverse multiplexing capability.

5.4 NetworkHub-Based Distribution via PRI

Conversely, the most expensive solution is to distribute the video bandwidth via PRI. Since this method is very expensive, it is reasonable only for sites that have to reach a codec that is more than 3000 wire feet from the NetworkHub. Figure 11 shows the configuration, which is similar in capability to the dual V.35/RS-366 line solution presented earlier. The NetworkHub dynamically switches B channels from the IXC PRI to the distribution PRI. The RemoteHub, a stripped-down version of the NetworkHub, then makes the bandwidth available to the codec through one or two V.35/RS-366 Interfaces. If there is more than one codec in a remote site, only one RemoteHub is required to service them both.

Advantages over 5.1:

- Can also have point-to-point videoconferences with sites using Video Telecom inverse multiplexing scheme,
- NetworkHub and RemoteHub can be managed together,
- PRI distribution line can be extended to almost any distance.

Disadvantage over 5.1:

- Requires a RemoteHub for each codec,
- Uses a PRI port on the NetworkHub for each codec.

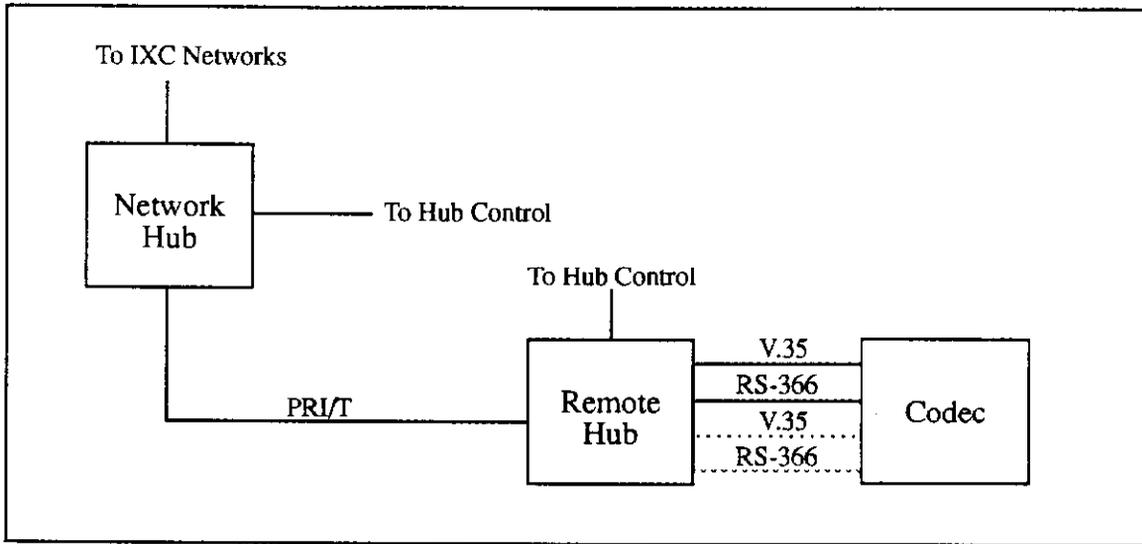


Figure 11. NetworkHub-Based Distribution via PRI

5.5 PBX-Based Distribution via BRI

Many PBXs or COs used as PBXs have the capability of distributing BRI ISDN from a direct PRI connection to an IXC. This configuration (see Figure 12) is identical to NetworkHub-based BRI distribution except that the PBX is taking the place of the NetworkHub. For sites that do not want higher bitrates than 128 Kbps and who already have a ISDN-capable PBX this is an attractive solution.

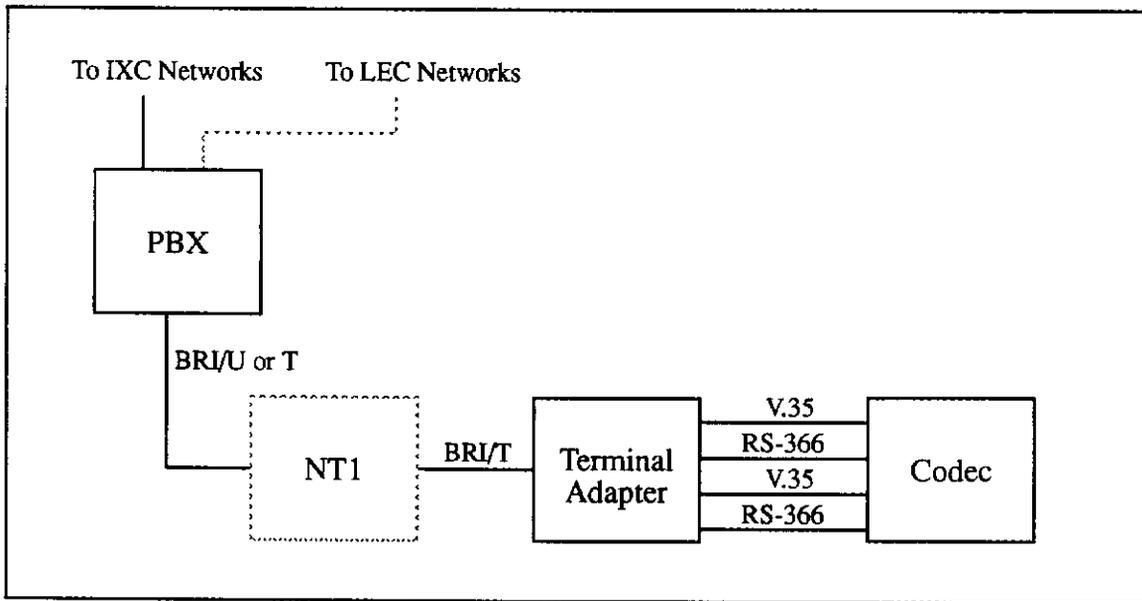


Figure 12. PBX-Based Distribution via BRI

Advantages over 5.1:

- Does not require a NetworkHub,
- Does not require HubExtenders,
- Uses existing wiring plant,
- Can have point-to-point videoconferences with sites using Video Telecom inverse multiplexing scheme.

Disadvantages over 5.1:

- Cannot be used for videoconferences at bitrates greater than 128 Kbps,
- Requires a terminal adapter for each codec,
- Must go through a gateway to communicate with core ERVN sites,
- Requires codec with inverse multiplexing capability.

5.6 LEC-Based Distribution via BRI

Finally, the simplest configuration (Figure 13) is available to those sites that have ISDN available from the LEC (or postal, telephone and telegraph administration (PTT) outside the US). Here, the LEC's CO acts as the distribution system. If the LEC offers BRI ISDN it should be just like ordering a new telephone line on site.

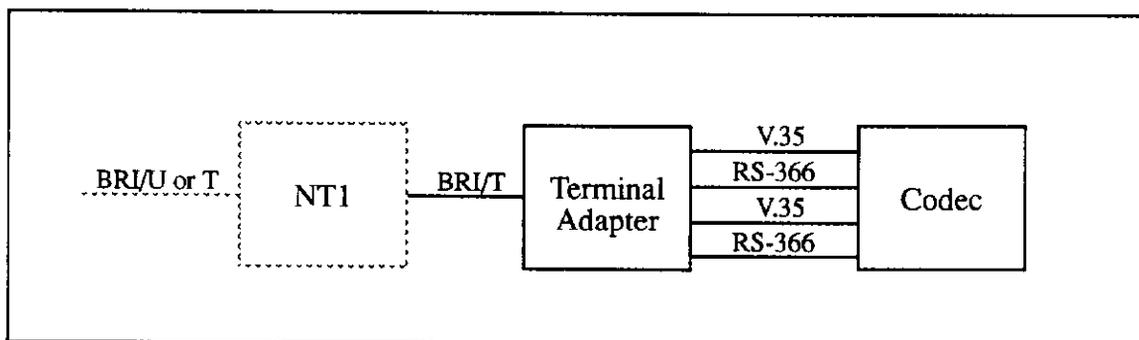


Figure 13. LEC- or PTT-Based Distribution via BRI

In the US, however, the LECs usually offer only crippled ISDN of one form or another which may limit bitrates to 112 Kbps. Sites that can only get 56 Kbps connections between COs may use this configuration to videoconference at 112 Kbps only.

Advantages over 5.1:

- Does not require a PRI connection,
- Does not require a NetworkHub,
- Does not require HubExtenders,
- Can access different IXC networks on a call-by-call basis with no PRI required,

- Can have point-to-point videoconferences with sites using Video Telecom inverse multiplexing scheme,
- May be able to use existing wiring plant.

Disadvantage over 5.1:

- Cannot be used for videoconferences at bitrates greater than 128 Kbps,
- Requires a terminal adapter for each codec,
- Depending upon LEC or PTT, may only be able to videoconference at 112 Kbps,
- Cannot use FTS-2000,
- Must go through a gateway to communicate with core ERVN sites,
- Requires codec with inverse multiplexing capability.

5.7 Hybrid Solutions

Sites that have several codecs or an MCU may need to use a hybrid of the architectures described. In this case, each one of the architectures can be thought of as a building block for the network. This allows each site some flexibility while still maintaining the integrity of the overall ERVN.

6. ORDERING EQUIPMENT AND SERVICES

Although each site will have to decide which configuration is appropriate in terms of cost and functionality, it is useful to provide recommendations for the majority of sites to follow. This will provide the highest degree of interoperability without having to use a gateway. It is expected that most ERVN videoconferences will be conducted with the following characteristics,

- Codecs will be CS-350s from Video Telecom,
- Codec bitrate will be 128 or 192 Kbps,
- Inverse multiplexing will be done in a NetworkHub using the Teleos scheme,
- Three or more sites will be involved,
- Multi-point bridging will be done by the VCC.

The configuration that matches this is described in Section 5.1, "NetworkHub-Based Distribution via a Single V.35/RS-366 Line". It is also possible to use the configuration described in Section 5.2, "NetworkHub-Based Distribution via Two V.35/RS-366 Lines" or the configuration described in Section 5.4, "NetworkHub-Based Distribution via PRI", but the cost to the site would be much higher. Throughout this section we will assume the configuration described in Section 5.1.

There are three major purchases required to be able to participate in video teleconferencing with other Energy research sites. These are the acquisition of the ISDN circuit, the purchase of an interface box to manipulate the ISDN "B" channels of the circuit, and the purchase of a videoconferencing system itself. Each will be discussed in some detail. Before equipment is ordered, the VCC should be consulted.

6.1 PRI Communications Circuit

Department of Energy (DOE) facilities should contact their designated area representative (DAR) and initiate the paperwork necessary to get an AT&T FTS-2000 PRI circuit installed. Non-DOE facilities should contact the National HEPnet Management (NHM) organization. NHM will assist in ordering a FTS-2000 PRI circuits for remote sites. The NHM Manager is Bill Lidinsky; he can be reached at +1 708 840-8067.

There are a couple of points to keep in mind as this circuit is ordered/installed.

- There is a \$3,000 installation charge for the circuit to be installed (to be reduced to \$2,000 as of October 1, 1992).
- There is a lead time of 120 working days for the installation of the circuit once the order is placed.
- There is no monthly lease fee for the circuit; all charges for the circuit are billed by the call (call costs vary with usage but the maximum cost is on the order of \$45 to \$65 per hour for 2 B channels).
- The party initiating the call gets the bill (if the SSC wants to conference with BNL and makes the ISDN call, the SSC will be charged for the call).
- AT&T FTS-2000 Primary Rate ISDN service provides a Channel Service Unit (CSU) to terminate the circuit which should be configured with extended super-frame (ESF) and B8ZS coding.
- Service should be ordered with "call-by-call selection" and all B channels active.
- A separate POTS (plain, old telephone service) line must be ordered for use by AT&T to dial-up their termination equipment for diagnostics.

Sites that are unwilling or unable to order FTS-2000 service can order commercial service from an IXC. All major carriers offer PRI ISDN access to their data networks, but since there are no interconnections between these networks, care should be taken in choosing a vendor. The VCC should approve of the vendor and have PRI service from that vendor as well so that they can provide gateway service. Finally, in the US, the CSU is considered customer premises equipment (CPE) and must be purchased separately.

6.2 PRI Circuit Termination Equipment

The NetworkHub systems are available in a variety of configurations, depending on the videoconferencing needs of the particular site. For sites that are going to use the FTS-2000 circuit only for videoconferencing and have only one or two videoconferencing systems, the NetworkHub Series 6000 Model 60 is recommended. Table 3 lists the equipment that should be ordered.

Sites that will have their NetworkHub and their codec co-located do not need any HubExtenders. Also, a site must order one STIU-C for each codec they have. Finally, a site that does not want any local management may opt not to get the LMO-C. These variations cause list pricing for a back-

Item	Description	List Price
VH7000-60	NetworkHub with 1 PRIU-S + 3 Slots	\$9,000
STIU-01	Sync. Terminal Interface Unit [2 V.35/RS-366 Ports]	3,000
I-MUX	Inverse Multiplexing Software for STIU	1,500
PRI/TI Cables	PRI Cables for PRI/T-US	95
STIU-C	Dual V.35/RS-366 Cable (15 ft.)	400
258-A	AMP to RJ45 Harmonica Adapter	50
LMO-C	Local Management Option - Color	1,895
HUB-X	1 Pair Hub Extenders with V.35/RS-366	2,500
COOP-1	Cooperative Maintenance 1 (per year)	10% of total
INSTALL	Installation	8% of equipment

Table 3: NetworkHub Itemized

bone site to range from \$17,000 to \$30,000 depending upon the number and location of codecs and the desire for local management.

Teleos Communications, Inc. markets directly and through dealers. They have sales offices in several U.S. locations and their home office is in Eatontown, New Jersey. They may be contacted at:

Teleos Communications Inc.
 2 Meridian Road
 Eatontown, New Jersey 07724
 phone: +1 908 389-5700

6.3 Videoconferencing Equipment

At the present time, the videoconferencing equipment in use at Energy Sciences research sites is all manufactured by Video Telecom, Inc. (Austin, Texas). It is a vendor proprietary solution; that is, it won't interoperate with equipment from Compression Labs, PictureTel, etc. The reason for this is that the CCITT H.261 codec interoperability standard has not yet been adopted; the standards committee is currently working on the issue of audio compression. When this problem is resolved and the H.261 standard is finalized with audio compression, all of the Video Telecom equipment will be upgraded to be compliant with the standard.

Consequently, an energy research site should purchase a CSR 425BK videoconferencing system manufactured by Video Telecom (see Table 4). The CSR 425BK videoconferencing system consists of the following:

- CS350 codec,

- dual 56/64 interface with RS-366 dialing,
- pan/tilt/zoom camera and motorized control,
- a document camera,
- Pen Pal Graphics,
- hand held remote control unit,
- two microphones,
- speaker,
- two Sony 25 inch color video monitors,
- roll-about enclosure,
- associated cables.

Item	Description	List Price
CS 425BK	CS 350 Codec and Associated Equipment	\$59,000
112-300	Dual 56/64 with 2 V.35 and 2 RS-366 Interfaces (for older codecs only)	2,185
MAINT	Maintenance on codec (after first year)	3,695

Table 4: Videoconferencing System Itemized List

Sites that already have CS 350 codecs must purchase an upgrade kit that allows the codec to place dual 56 or 64 Kbps calls. This upgrade may involve shipping the codec back to the factory. A loaner codec can be arranged from Video Telecom during the upgrade.

The CS 350 is an older codec that is not being manufactured any more, so why recommend it? The answer is compatibility. The new MediaMax's do not support conferences with the CS 350s that many energy research sites have now. When the new model of the MediaMax that supports such backward compatibility is available all sites should upgrade. The SSCL has negotiated a deal that will allow any energy research site to upgrade its CS 350 to a MediaMax for \$3,500.

Video Telecom markets only through dealers. Their main office is in Austin, Texas. VTC can be reached at:

Video Telecom Corporation
 1901 W. Breaker
 Austin, TX 78758
 phone: +1 512 834-2700

6.4 Alternate Equipment And Services

Some sites may not be able or willing to buy a NetworkHub or order an FTS-2000 PRI. Sites that plan only occasional participation in the ERVN or international sites that have good ISDN con-

nectivity from their PTT may fall into this category. Conferences from such sites are expected to have the following characteristics:

- Codecs will be CS-350s from Video Telecom,
- Codec bitrate will be 128 Kbps,
- Inverse multiplexing will be done in the codec using the Video Telecom scheme,
- Three or more sites will be involved,
- The VCC will act as a gateway.

These sites would use the configuration described in Section 5.5, "PBX-Based Distribution via BRI" or the configuration described in Section 5.6, "LEC-Based Distribution via BRI". Like the set-up detailed in Section 6, there are three major expenses, the acquisition of the ISDN circuit, the network termination equipment, and the videoconferencing system itself. We will discuss each of these in detail.

6.5 Alternate BRI Communications Circuit

- There are two types of BRI service that may be used. The first is supplied by the LEC or PTT serving the site. Ordering this service should be equivalent to ordering a telephone, but many telephone companies have little experience with ISDN. BRI service should be ordered with the following characteristics:
 - 2 B channels available over 1 BRI line (some LECs only offer 1B+D),
 - full 64 Kbps B channel connections between COs (many LECs only offer 56 Kbps connections between COs),
 - ability to connect to an IXC ISDN network supported by the VCC (some LECs only allow access within their area).
 - terminal type compatible with terminal adapter to be used.

Most LECs do not offer unrestricted 64 Kbps service between COs due to deficiencies in inter-office signalling. In this case it is recommended that the site use another configuration that does not rely on the LEC. If no other configuration is viable, then the site can order a BRI circuit supporting 56 Kbps rate-adapted circuit-switched data between COs. Using this type of service, however, will force everyone involved in a videoconference with this site to use 112 Kbps.

The second type of BRI service that may be used is that supplied by an on-site PBX or CO used as a PBX. In this case, the site should order a FTS-2000 or commercial PRI as described in Section 6.1. A BRI should then be run from the PBX with the following characteristics:

- 2 B channels active,
- unrestricted 64 Kbps circuit-switched data on both B channels,
- terminal type compatible with terminal adapter to be used.

6.6 Alternate BRI Circuit Termination Equipment

Terminating a BRI circuit is much simpler than terminating a PRI. If the circuit is delivered via a U (two-wire) interface a network termination unit (NTU) must be purchased to provide isolation and conversion to a T (four-wire) interface. If the serving CO or PBX is close by, the circuit may be delivered via a T interface and no NTU is required. Since international standards are not yet fixed, the NTU must be matched to the serving CO or PBX. For example the AT&T NT1U-200 works with the AT&T 5ESS CO and costs about \$300. AT&T sells only through dealers, typically the local service provider.

Next, a terminal adapter (TA) provides conversion of ISDN bearer services and signalling to the more familiar V.35/RS-366 interface. A TA should be purchased that provides a V.35 and RS-366 interface for each of the B channels. Like the NTU, the TA must be matched to the serving switch. For example the Gandalf TA-1 works with the AT&T 5ESS and the Northern Telecom DMS-100 and sells for about \$1,595. Gandalf can be reached at:

Gandalf Data, Inc.
1020 S. Noel Ave.
Wheeling, IL 60090
phone: +1 800 426-3253
or +1 609 424-9400

7. CONCLUSION

The ERVN is rapidly changing from a pilot project into a full-scale production system. The investment in Video Telecom equipment will be protected, but the interconnect between the systems will be provided by digital switched service instead of dedicated links. In addition, a central VCC will provide scheduling and management of the network. With these enhancements, adding new sites and enhancements existing sites will be easily made.

8. ACKNOWLEDGEMENTS

Many people have contributed to the design of the ERVN as documented in this paper, especially Ari Ollikainen of NERSC and Wayne Gore of the SSCL. In addition, Andy Kaufmann of Teleos has been patient and knowledgeable. Bill Lidinsky was instrumental in getting this paper started and has reviewed many versions.

9. GLOSSARY

ATM - Asynchronous Transfer Method. This is a derivative of fast packet switching technology and is a name assigned by CCITT to this technology during the development of broadband ISDN standards to distinguish it from Synchronous Transfer Mode (STM) technology. Fast packet switching can handle variable length packets, but ATM uses short fixed-size packets called cells. Each cell consists of a 5-byte header and a 48-byte information field. Short, fixed length cells are chosen to minimize delay variability and to simplify the design of the cell switch and buffer management.

“B” channel - The “B” channel is the basic user communications channel for ISDN services. It is used to carry digital data, pulse code modulated (PCM) encoded digital voice, etc. at a rate of 64,000 bits per second.

Bandwidth Aggregation - See “Inverse Multiplexing”.

BONDING - Bandwidth On Demand Interoperability Group. A group of inverse multiplexer vendors formed to establish standards for the inter-operation of inverse multiplexers.

BRI - Basic Rate Interface. The ISDN Basic Rate Interface (2B+D) consists of two “B,” bearer, channels and one “D,” control, channel. Each “B” channel transfers data at bitrates up to 64,000 bits per second full-duplex. The “D” channel transfers data at bitrates up to 16,000 bits per second full-duplex.

BRI/T - Basic Rate Interface/T-interface. This is a four wire interface, normally delivered via an RJ-45 jack.

BRI/U - a Basic Rate Interface/U-interface. This is a two wire interface, normally delivered via a RJ-11 jack.

CCITT - Comite' Consultatif International Telegraphique et Telephonique (the International Telegraph and Telephone Consultative Committee). This is an international organization that decides upon recommended communication protocol standards.

CODEC - COder/DECoder. This is the crucial component of a videoconferencing system. The codec takes the audio and video analog signals from a site and converts each to a compressed digital stream. The streams are then combined and sent over the communications link. The codec also accepts the combined digital stream from a remote codec, decompresses and converts it to its component audio and video signals and to drive the site speaker and video monitor.

CO - Central Office Switch. This is the computing and switching system that routes calls in the public switched telephone network. Each ISDN BRI or PRI line is connected to a central office switch. The subscriber database is contained in the central office switch.

CSU - Channel Service Unit. This is a component of customer premises equipment used to terminate a digital circuit (such as a DDS or T1) at the customer site; it performs certain line conditioning functions, ensures compliance with FCC rules, and responds to loopback commands from the central office. It also ensures proper "ones" density in the transmitted bit stream and corrects bipolar violations for transmission.

"D" Channel - The "D" channel is the circuit control channel for ISDN services. It carries signalling information to control circuit-switched calls on associated "B" channels of the user interface. The "D" channel can also be used for packet-switching or low bitrate (e.g. - 1000 bps) telemetry when no signalling information is present.

DDS - Digital Data Service. This is a private line digital service offered intraLATA by Bell Operating Companies and interLATA by AT&T communications. The rates are typically 2.4, 4.8, 9.6 and 56 Kbps, and are part of the services tariffed by AT&T as the Accunet family.

DPE - Data Path Extender. This is a component of customer premises equipment used to extend the physical (wire) distance over which some other electronic subsystem will operate. Such units are often termed line drivers or limited distance modems.

DSU - Data Service Unit. This is a component of customer premises equipment used to interface to a digital circuit (e.g. - DDS or T1) combined with a channel service unit (CSU); this converts a customer's data stream to bipolar format for transmission.

DVBX - Digital Video Branch Exchange. See "MCU."

E1 - European version of a T1 consisting of a circuit running at 2.048 Mbps. It can be divided into 30 64 Kbps bearer channels and two 64 Kbps signalling channels for ISDN.

H.261 - A set of standards developed by the International Telegraph and Telephone Consultative Committee (CCITT) specifying video codec operation at rates of $p \times 64$ Kbps (where p is an integer in the range 1 to 30).

I.451/Q.931 - A set of standards developed by the International Telegraph and Telephone Consultative Committee (CCITT) specifying the manner in which an ISDN subscriber interacts with the ISDN network for call setup and control.

Inverse Multiplexing - A technique used to take a number of low bitrate communications circuits (e.g. - an ISDN "B" channel at 64 Kbps) and aggregate them into a higher bitrate channel which acts as a single communications circuit. Also known as "bandwidth aggregation."

ISDN - Integrated Services Digital Network. This is an evolving worldwide telecommunications service that uses digital transmission and switching technology to support both voice and digital data communications.

IXC - Inter-eXchange Carrier. In the US, this is a company providing long distance communications services between Local Exchange Carriers (LEC's). Examples of such companies are AT&T, MCI, Sprint, and WITel. Internationally, the PTT provides IXC-type service.

LATA - Local Access Transport Area. A geographic area encompassing a metropolitan or rural area. Most calls within a LATA are carried by the LEC.

LEC - Local Exchange Carrier. In the US this is a local telephone company, either one of the Bell Operating Companies or one of the 1400+ independent local telephone companies. In most other countries the LEC and IXC are combined.

MCU - Multi-point Control Unit. This is a electronic subsystem used in a video conference where three (or more) codecs participate; it will arbitrate which picture is displayed on the conference monitor, usually employing a voice activated switching technique. Also known as a "VBX."

Microsoft Windows - Microsoft Windows (TM) is the graphical user interface most frequently run atop MS/DOS. It provides a multitasking icon/menu environment to personal computer users.

MS/DOS - Microsoft Disk Operating System. This is the operating system environment most frequently in use on personal computers that have an Intel x86 processor chip as the compute engine.

PCM - Pulse Code Modulation. Method of converting an analog voice signal to digital.

PBX - Private Branch Exchange. This is a private switching system, usually serving an organization such as a business or government agency, and usually located on the customer's premises.

Px64 - See "H.261."

POTS - Plain Old Telephone Service. Traditional analog voice service.

PRI - Primary Rate Interface. The ISDN Primary Rate Interface (23B+D) consists of twenty-three "B" channels and one "D" channel. Each "B" channel transfers data at bitrates up to 64,000 bits per second full-duplex. The "D" channel transfers data at bitrates up to 64,000 bits per second full-duplex.

PSTN - Public Switched Telephone Network. This is the portion of the total network that supports public switched telephone network services. It provides the capability of interconnecting virtually any home or office in the country with any other.

PTT - Postal, Telephone and Telegraph. This term is used to refer to government agencies responsible for regulating and providing telecommunications. In most non-US countries, telephone, telegraph and postal service are provided by the same governmental agency.

RS-366 - A physical layer standard developed by the Electronic Industries Association (EIA) which specifies the interface between data terminal equipment (DTE) and automatic calling equipment. RS-366 specifies mechanical, functional, and procedural characteristics.

RS-449 - A set of physical layer standards developed by the Electronic Industries Association (EIA) intended to replace the RS-232C standard. RS-449 specifies mechanical, functional, and procedural characteristics.

Switched 56 - The most common form of data service offered by LECs. It provides 56 Kbps of bandwidth and is delivered via normal twisted pairs.

TA - An (ISDN) Terminal Adapter. This is an electronic subsystem that maps a non-ISDN terminal, personal computer, multiplexer, modem, etc. into an ISDN device. In most cases, the adaption is to the basic rate (2B + D) interface.

T1 - A T1 is a communications circuit running at 1.544 Mbps. It is often broken into 23 64 Kbps voice channels and a 64 Kbps signaling channel. In data circuits it is often used as a single clear channel.

UTP - Unshielded Twisted Pair. This is the normal building wiring for telecommunications services. It consists of two insulated copper conductors that are arranged in a regular spiral pattern.

V.35 - An International Telegraph and Telephone Consultative Committee (CCITT) 48 Kbps leased line modem recommendation. It is also used to refer to the "M" series 34 pin cable/connected assembly used to connect communications equipment to a CSU/DSU.

VBX - Video Branch Exchange. See "MCU."