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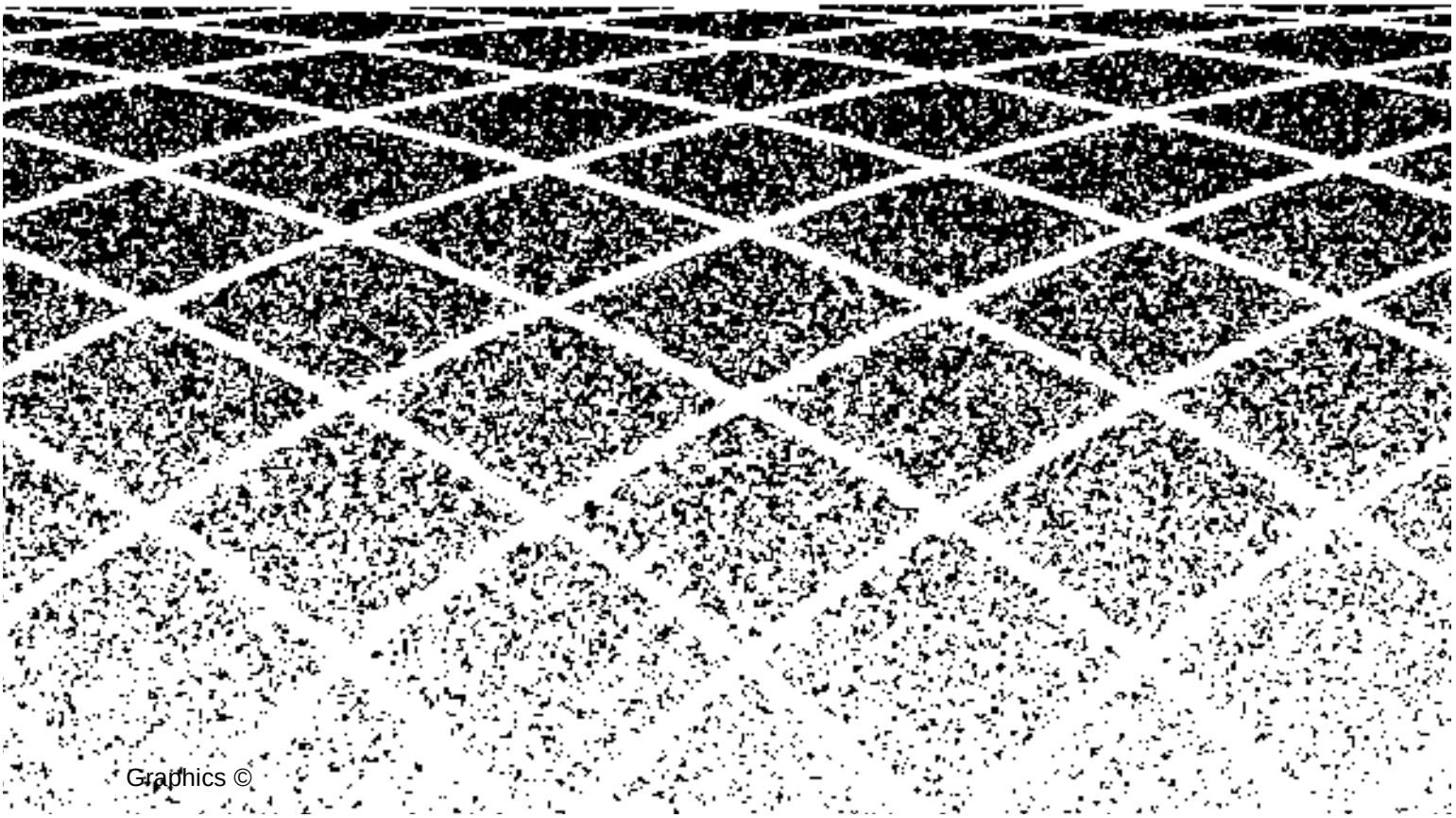
Multi-Point Control Unit
R3.0 Preconfigured
Single Conference



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Preconfigured Single Conference MCU

Imagine that you are a supervisor in a branch of a large company and that you have a proposal you'd like to share as soon as possible with members of the company at another site. Imagine, too, that you have prepared a set of still-frame graphs supporting your proposal. Now, imagine your disappointment when you schedule a telephone conference call and suddenly realize that nobody in the conference except you will be able to examine the graphs today unless you can somehow fax them.....

A multimedia conferencing system can help solve your company's or organization's information movement challenges. This is true for even the most basic of such systems that are in any way reputable. The AT&T Preconfigured Single Conference MCU is a basic Multipoint Control Unit (MCU) that is inexpensive, reliable and easy to use. And the system is flexible—it includes an evolving system architecture that is designed to meet your company's changing needs as they evolve. New equipment, applications, and features can be added easily in the future without requiring huge investments. As a basic system, the Preconfigured Single Conference MCU sets a new price point for the MCU industry.

We feel that, once you examine what the Preconfigured Single Conference MCU has to offer, you will agree that using this system can be "better than being there"—because the system allows you to get the right people together to work on the right information and, therefore, make faster and better-informed business decisions.

Building on a Solid Foundation

The Preconfigured Single Conference MCU derives its processor, memory, backplace, cabinet, and software platform from the AT&T DEFINITY® Generic 3s switch. As such, the system has many of the flexible communications capabilities provided by this switch.

The Preconfigured Single Conference MCU is basic, but flexible—it supports up to eight ports, but it allows you to add more ports to meet your company's growing requirements. In fact, you can build on your system to include as many as 64 ports. The more ports you have in your system, the more participants you can include in a conference. This flexibility protects your investment in the Preconfigured Single Conference MCU.

Basic Components

The AT&T Preconfigured Single Conference MCU transmits audio and video between endpoints within a multimedia conference. Endpoints can include group systems or personal computers that are equipped to handle multimedia calls. The Preconfigured Single Conference MCU and its corresponding endpoints are linked via either a public communications network, such as the AT&T Switched Digital Services (SDS), or a Private Branch Exchange (PBX) capable of supporting switched digital data services. Robbed-bit trunks are used to make these connections. For customers in Australia and the United Kingdom, an ISDN interface that provides 64k channels is supported.

The Preconfigured Single Conference MCU is available in two configurations: four multimedia ports, or six multimedia ports plus two Audio Add-on ports. The system supports one multimedia conference at a time at a 2B-channel, 112k bandwidth. The audio portion of the conference can support up to four or eight speakers simultaneously, depending on the configuration.

The primary components of the system are housed in an Enhanced Single-Carrier Multimedia Server Module (ESCC-MSM). The ESCC-MSM contains a control complex and a port complex, each of which includes the appropriate circuit packs for system operation.

Basic Operation

A Preconfigured Single Conference MCU conference is always “on” — there is no need to reserve this type of conference.

You can join this type of conference by placing a “Meet-Me” call. To place a “Meet-Me” call, you dial from a voice terminal the appropriate “Meet-Me” number, which is pre-administered by a technician at the AT&T Video Technical Center (VTC). The MCU for the conference is assigned up to eight “Meet-Me” numbers; this allows up to eight endpoints to connect to the unit simultaneously.

Once you place the call, the unit processes the “Meet-Me” call digits, maps them to an MCU-extension, and assigns the MCU-extension to you, the conferee. Your call is then terminated on an MCU port. An MCU port is a collection of hardware and software resources that allow an endpoint using a supported transfer rate to connect to the MCU and participate in a conference. Once the conference is completed, your port is put back into a “pool” and your MCU-extension is again available for use.

Voice-Activated Switching Conference Mode

A conference mode determines what each conferee sees and/or hears at a given moment during the conference. The Voice-Activated Switching conference mode is part of the Preconfigured Single Conference MCU functionality and requires no special features from the endpoint. For this conference mode, the current speaker sees the previous speaker and is seen by all the other conferees.

For example, consider a four-party video conference involving the following endpoints: “Site A,” “Site B, ” “Site C,” and “Site D.” Assume that “Site B” is the first speaker in the conference. Then assume that “Site B” stops talking and “Site A” begins to speak. As “Site A” begins to speak, the voice-activated switching function displays “Site A” to the other three endpoints. At the same time, “Site B” (the previous speaker) is displayed on the screen at “Site A” (the current speaker).

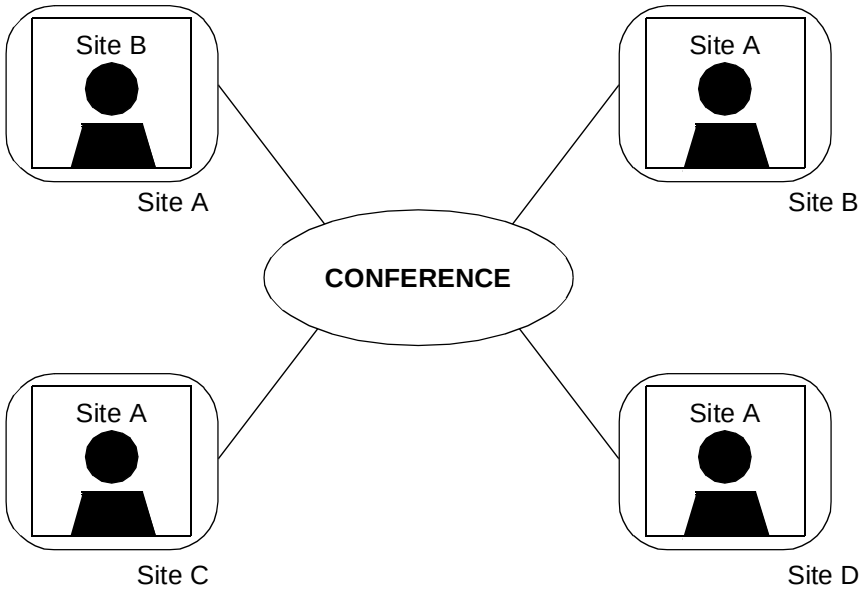

Figure 1. Speaker = Site A, Previous Speaker = Site B

The following figure illustrates what happens when the conferee at “Site A” stops talking and the conferee at “Site B” starts to talk. In this case, “Site B” (the current speaker) sees “Site A” (the previous speaker), and all the other endpoints see Site B.

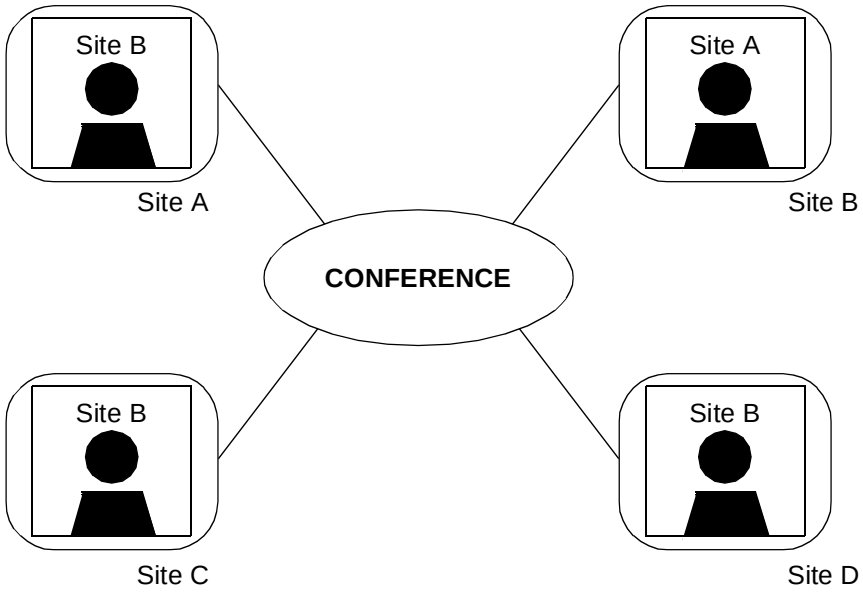

Figure 2. Speaker = Site B, Previous Speaker = Site A

Similarly, the following figure illustrates what happens when “Site B” stops talking and “Site D” begins to speak. In this case, “Site D” (the current speaker) sees “Site B” (the previous speaker), and all the other endpoints see “Site D.”

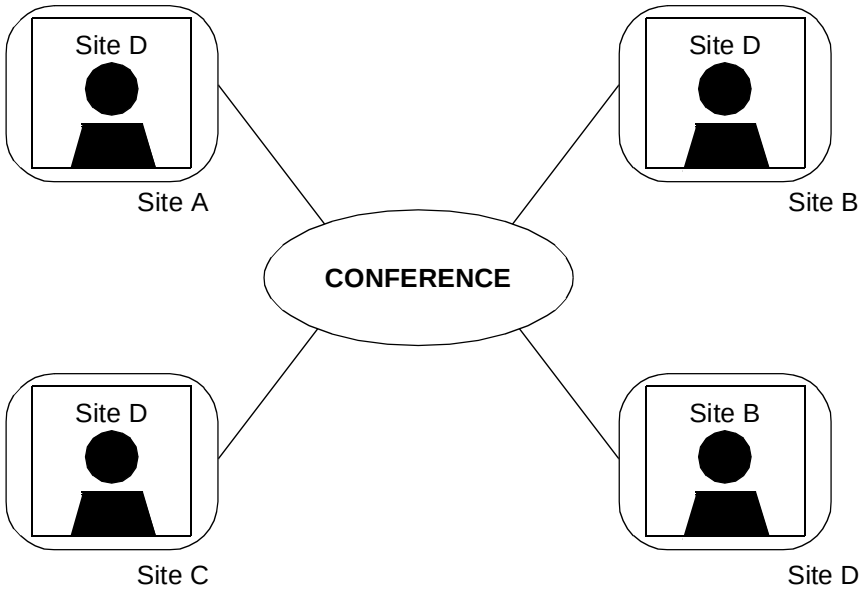

Figure 3. Speaker = Site D, Previous Speaker = Site B

Voice-Activated Switching is ideal for meetings in which all conferees are expected to share information.

Notification Tones

Notification tones are used during a conference to alert conferees about the status of a conference. Each tone has a different length and frequency so it can be distinguished from each other. Tones are selected on a per-conference basis at the time a conference is reserved.

Entry and Exit Tones

When a conferee joins a conference, an entry tone alerts other conferees already on the conference that another endpoint has joined. Similarly, when a conferee disconnects from a conference, an exit tone notifies the other conferees that an endpoint has dropped. Entry and exit tones are a customer option.

Warning Tone

This tone sounds when only 10 minutes remain in a conference. This tone gives advance notice to conferees to either conclude the conference or request additional time to extend the conference.

Monitoring System Usage

The Preconfigured Single Conference MCU includes several features that allow you to monitor system performance and security. For example, Call Detail Recording (CDR) allows you to generate reports for the conference services provided by the system. These reports contain information about the calls that come into the Preconfigured Single Conference MCU. This information may include the access codes used to place calls and several condition codes that provide additional information about the calls.

CDR can help detect hackers or unauthorized call attempts. It also allows you to establish a billing tool for use of your system.

You can use a local printer as the output device for this feature. The output device can be connected to the Preconfigured Single Conference MCU via the standard RS232 interface provided by the system.

Maintaining the System

AT&T is committed to ensuring that your system is properly maintained. As part of this commitment, AT&T technicians are available to assist you. If you are a customer in the US or Canada, technicians at the AT&T Video Technical Center

(VTC) are available. If you are a customer in the United Kingdom or Australia, technicians at the Global Technical Access Center (GTAC) are available to assist you. VTC and GTAC technicians can service alarms by dialing into your Preconfigured Single Conference MCU and performing the appropriate diagnostics.

These technicians are also available to take any phone calls concerning system problems. To consult the VTC, dial 800 242-2121. To consult the GTAC, dial 303 538-4666.

Multimedia Conferencing Strengths for Your Company

The strengths of the AT&T Preconfigured Single Conference MCU make it the premier “basic” multimedia conferencing system. Let’s review some of these strengths now and also consider a few more—and how they can benefit your company.

Natural Video Switching

The Preconfigured Single Conference MCU has a sophisticated speech detection algorithm, which is very helpful in conferences that are set up to switch to the video of the speaker. This algorithm can detect when the current speaker in the conference stops talking and another speaker starts to talk. Moreover, the algorithm ignores any background noise. As a result, conferees see natural video switching during the conference.

System Security

The system allows you to protect it from hackers via the Call Detail Recording feature. Also, the system provides additional security by sending a tone to all conference participants whenever an endpoint joins or leaves the conference.

Video Still-Image Transfer

The Preconfigured Single Conference MCU allows you to send H.261 Annex D high-resolution video still-images to the other endpoints in the conference. These endpoints must support the ITUT-T H.261 Annex D Still-Image capability.

Audio Capability

The Preconfigured Single Conference MCU supports several audio algorithms, including G.711 and G.728. G.711 uses 56k/64k bandwidth and provides toll-quality audio. G.728 uses 16k bandwidth and provides audio quality that is comparable to G.711. Using G.728 allows more bandwidth to be allocated to video and, therefore, enhances the video quality of the conference.