

# The Remote Media Immersion System\*

Internet Meets the Future Home Theater

Roger Zimmermann, Chris Kyriakakis, Cyrus Shahabi, Christos Papadopoulos,  
Alexander A. Sawchuk, Ulrich Neumann  
Integrated Media Systems Center  
University of Southern California  
Los Angeles, California 90089

[rzimmerm, ckyriak, cshahabi, christos, sawchuk, uneumann]@usc.edu

## Abstract

The Remote Media Immersion (RMI) system is the result of a unique blend of multiple cutting-edge media technologies to create the ultimate digital media delivery platform. The main goal is to provide an immersive user experience of the highest quality. RMI encompasses all end-to-end aspects from media acquisition, storage, transmission up to their final rendering. Specifically, the Yima streaming media server delivers multiple high bandwidth streams, transmission error and flow control protocols ensure data integrity, and high-definition video combined with immersive audio provide highest quality rendering. The RMI system is operational and has been successfully demonstrated in small and large venues. Relying on the continued advances in electronics integration and residential broadband improvement, RMI demonstrates the future of on-demand home entertainment.

## Keywords

Streaming media, immersive environments, high-definition video, immersive audio, clustered server.

## 1 Introduction

The charter of the Integrated Media Systems Center (IMSC) at the University of Southern California

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(USC) is to investigate new methods and technologies that combine multiple modalities into highly effective, immersive technologies, applications and environments. One of the results of these research efforts is the Remote Media Immersion (RMI) system. The goal of the RMI is to create and develop a complete aural and visual environment that places a participant or group of participants in a virtual space where they can experience events that occurred in different physical locations. RMI technology can effectively overcome the barriers of time and space to enable, on demand, the realistic recreation of visual and aural cues recorded in widely separated locations [MNNS99].

The focus of the RMI effort is to enable the most realistic recreation of an event possible while streaming the data over the Internet. Therefore, we push the technological boundaries much beyond what current video-on-demand or streaming media systems can deliver. As a consequence, high-end rendering equipment and significant transmission bandwidth are required. However, we trust that advances in electronics, compression and residential broadband technologies will make such a system feasible first in commercial settings and later at home in the not too distant future. Some of the indicators that support this assumption are, for example, that the next generation of the DVD specification calls for network access of DVD players [Smi03]. Furthermore, Forrester Research forecasts that almost 15 per cent of films will be viewed by “on-demand” services rather than by DVD or video by 2005 [Ric03a]. The infrastructure necessary for these services is gradually being built as is demonstrated in Utah, where 17 cities are planning to construct an ultra-high speed network

for both businesses and residents [Ric03b].

The RMI project integrates several technologies that are the result of research efforts at IMSC. The current operational version is based on four major components that are responsible for the *acquisition, storage, transmission, and rendering* of high quality media.

- **Acquisition of high-quality media streams.**

This authoring component is an important part of the overall chain to ensure the high quality of the rendering result as experienced by users at a later time. As the saying “garbage in, garbage out” implies, no amount of quality control in later stages of the delivery chain can make up for poorly acquired media. In the current RMI version, authoring is an offline process and involves its own set of technologies. Because of space constraints, we will not focus on this part.

- **Real-time digital storage and playback of multiple independent streams.**

The *Yima* [SZFY02] Scalable Streaming Media Architecture provides real-time storage, retrieval and transmission capabilities. The Yima server is based on a scalable cluster design. Each cluster node is an off-the-shelf personal computer with attached storage devices and, for example, a Fast or Gigabit Ethernet connection. The Yima server software manages the storage and network resources to provide real-time service to the multiple clients that are requesting media streams. Media types include, but are not limited to, MPEG-2 at NTSC and HDTV resolutions, multichannel audio (e.g., 10.2 channel immersive audio), and MPEG-4.

- **Protocols for synchronized, efficient real-time transmission of multiple media streams.**

A selective data retransmission scheme improves playback quality while maintaining real-time properties. A flow control component reduces network traffic variability and enables streams of various characteristics to be synchronized at the rendering location. Industry standard networking protocols such as Real-Time Protocol (RTP) and Real-Time Streaming Protocol (RTSP) provide compatibility with commercial systems.

- **Rendering of immersive audio and high resolution video.**

Immersive audio is a technique developed at IMSC for capturing the audio environment at a remote site and accurately reproducing the complete audio sensation and ambience at the client location with full fidelity, dynamic range and directionality for a group of listeners (16 channels of uncompressed linear PCM at a data rate of up to 17.6 Mb/s). The RMI video is rendered in HDTV resolutions (1080i or 720p format) and transmitted at a rate of up to 45 Mb/s.

In this report we detail some of these components and the techniques that are employed within each. The hope is that our advances in digital media delivery will enable new applications in the future, be that in the entertainment sector (sports bars, digital cinemas, and eventually the home theater), distance education, or others. We will focus mainly on the transmission and rendering aspects.

## 2 Overview of Components

The overall objective of the RMI research endeavor is to achieve the best possible quality at each rendering location for a group of participants. Group sizes may range from a single person or family at home to a large venue seating hundreds. For the visual streams we decided that we required at least high-definition (HD) resolution as defined by ATSC<sup>1</sup>. The highest quality ATSC modes are either 1920 × 1080 pixels at an interlaced frame rate of 29.97 per second, or 1280 × 720 pixels at a progressive frame rate of 59.94 per second. Details of the video playback are discussed in Section 4. For the audio rendering we rely on the immersive audio technology developed at IMSC which utilizes a 10.2 channel playback system. The rendering capabilities of immersive audio goes much beyond current stereo and 5.1 channel systems and more details are contained in Section 4.1. The combination of 10.2 channels of immersive audio and high-definition video is the next step in audio-visual fidelity.

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<sup>1</sup>Advanced Television Systems Committee, <http://www.atsc.org>.



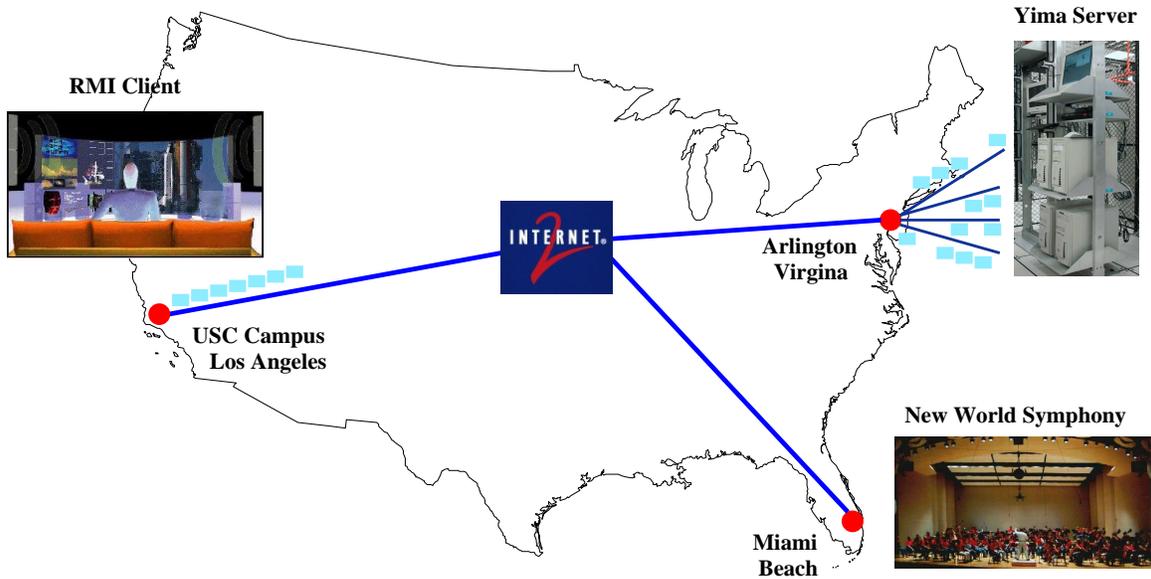


Figure 2: The RMI test environment. A server is located at the Information Sciences Institute in Arlington, VA, while the rendering is performed in Los Angeles.

quence [BGMJ94], or in a *random* manner [SM98]. Yima uses a pseudo-random data placement in combination with a deadline-driven scheduling approach. This combination enables short startup latencies and can easily support multimedia applications with non-sequential data access patterns including variable bit rate (VBR) video or audio, and interactive operations such as pause and resume, etc. It also enables efficient data reorganization during system and storage scaling [SMRN00]. For the delivery of media data we take full advantage of this high performance storage layout.

### 3.1 Error Control with Selective Retransmissions from Multiple Server Nodes

One of the characteristics of continuous media streams is that they require data to be delivered from the server to a client location at a predetermined rate. This rate may vary over time for streams that have been compressed with a variable bit rate (VBR) media encoder. VBR streams enhance the rendering quality, however they will generate bursty traffic on a packet switched network such as the Internet. This in turn can easily lead to packet loss due to congestion. Such data loss adversely affects compressed audio and video streams because much of the temporal or

spatial redundancy in the data has already been removed by the compression algorithm. Furthermore, important data such as audio/video synchronization information may get lost that will introduce artifacts in a stream for longer than a single frame. As a result it is imperative that as little as possible of a stream's data is lost during the transmission between the server and a rendering location.

The Yima cluster architecture takes advantage not only of the distributed storage resources among the multiple nodes, but also of the multiple network connections that link all the nodes together. Media data is transmitted via the real-time protocol (RTP) encapsulated in connection-less UDP datagrams. To avoid traffic bottlenecks, each node transmits the data blocks that it holds directly to the clients via RTP. Hence, each client will receive RTP data packets from each server node within the cluster. Note that the current Internet infrastructure was not designed for streaming media and provides only best-effort packet delivery. Therefore, RTP/UDP datagrams are not guaranteed to arrive in order or at all. Reordering can easily be achieved with the help of a global sequence number across all packets, but information loss requires special provisions if the quality of the rendered streams at the receiving side should be acceptable.

One possible solution is the use of forward error correction (FEC). With this method, the server continuously adds redundant information to the stream that aids the receiver in the reconstruction of the original information if data is corrupted or lost during transmission. Because of its preemptive nature, FEC can add significant overhead that consumes additional bandwidth even when it is not needed. With RMI we are transmitting some streams that require in excess of 45 Mb/s bandwidth [Tau02]). In that case, retransmission-based error control (RBEC) is an attractive option. RBEC has been shown to be an effective solution for streaming media applications that employ a playout buffer at the client side [PP96].

A central question arises when data is randomly stored across multiple server nodes and retransmission-based error control is employed: *When multiple servers deliver packets that are part of a single stream, and a packet does not arrive, how does the client know which server node attempted to send it?*

In other words, it is not obvious where the client should send its request for retransmission of the packet. There are two general solutions to this problem. First, the client can broadcast the retransmission request to all server nodes, or second, it can compute the server node to which it issues the retransmission request.

**Broadcast Retransmissions.** With the broadcast approach, all server nodes receive a packet retransmission request. Please note that the request broadcasting in this scenario can be well targeted to include all the server nodes, but no other computers. From observing the RTP/UDP packet header source IP address, the client can easily establish the complete set of server nodes. Once a server receives a request it checks whether it holds the packet, and either ignores the request or performs a retransmission. Consequently, this approach wastes network bandwidth and increases server load.

**Unicast Retransmissions.** The second, more efficient and scalable method of sending retransmission requests requires that the unique server node that holds the missing packet be identified. This could be accomplished in several ways. For example, the

client could reproduce the pseudo-random number sequence that was originally used to place the data across multiple server nodes. This approach has several drawbacks. First, identical algorithms on both the clients and the servers must be used at all times. If the server software is upgraded then all clients must be upgraded immediately too. The logistics of such an undertaking can be daunting if the clients are distributed among thousands of end users. Second, during scaling operations the number of server nodes or disk drives changes and hence new parameters need to be propagated to the clients immediately. Otherwise, the server nodes will be misidentified. Third, if for any reason the client computation is ahead or behind the server computation (e.g., the total number of packets received does not match the number of packets sent), then any future computations will be wrong. This could potentially happen if the client has only a limited memory and packets arrive sufficiently out-of-sequence.

A more robust approach is as follows. The client determines the server node from which a lost RTP packet was intended to be delivered by detecting gaps in node-specific packet sequence numbers. We term these *local* sequence numbers (LSN) as opposed to the *global* sequence number (GSN) that orders all packets. Although this approach requires packets to contain a node-specific sequence number along with a global sequence number, the clients require very little computation to identify and locate missing packets.

We have implemented this technique and evaluated it with an extensive set of experiments across LAN and WAN environments. The results show that the method is feasible and effective. Further details may be found in [ZFNS03].

### 3.2 Client-Server Adaptive Flow Control

RMI relies on the efficient transfer of multimedia streams between a server and a client. These media streams are captured and displayed at a predetermined rate. For example, video streams may require a rate of 24, 29.97, 30, 59.94, or 60 frames per second. Audio streams may require 44,100 or 48,000 samples per second. An important measure of quality for such multimedia communications is the precisely timed playback of the streams at the client location.

Achieving this precise playback is complicated by the popular use of variable bit rate (VBR) media stream compression. VBR encoding algorithms allocate more bits per time to complex parts of a stream and fewer bits to simple parts to keep the visual and aural quality at near constant levels. For example, an action sequence in a movie may require more bits per second than the credits that are displayed at the end. As a result, different transmission rates may be required over the length of a media stream to avoid starvation or overflow of the client buffer. As a contradictory requirement we would like to minimize the variability of the data transmitted through a network. High variability produces uneven resource utilization and may lead to congestion and exacerbate display disruptions.

The focus of the RMI flow control mechanism is on achieving high quality media playback by reducing the variability of the transmitted data and hence avoiding display disruptions due to data starvation or overflow at the client. We designed a novel technique that adjusts the multimedia traffic based on an end-to-end rate control mechanism in conjunction with an intelligent buffer management scheme. Unlike previous approaches, we consider multiple signaling thresholds and adaptively predict the future bandwidth requirements. With this *Multi-Threshold Flow Control* (MTFC) scheme, VBR streams are accommodated without a priori knowledge of the stream bit rate. Furthermore, because the MTFC algorithm encompasses server, network and clients, it adapts itself to changing network conditions. Display disruptions are minimized even with few client resources (e.g., a small buffer size). The MTFC protocol is currently implemented in our Yima [SZFY02, ZFS<sup>+</sup>01] streaming media server and clients. This allows us to measure and verify its effectiveness in an end-to-end application such as RMI.

The design of a high performance rate control algorithm was motivated in part by our continuous media server implementation. We required an adaptive technique that could work in a real-world dynamic environment with minimal prior knowledge of the multimedia streams to be served. We identified the following desirable characteristics for the algorithm:

- **Online operation:** Required for live streaming and also desirable for stored streams.

- **Content independence:** An algorithm that is not tied to any particular encoding technique will continue to work when new compression algorithms are introduced.
- **Minimizing feedback control signaling:** The overhead of online signaling should be negligible to compete with offline methods that do not need any signaling.
- **Rate smoothing:** The peak data rate as well as the number of rate changes should be lowered compared with the original, unsmoothed stream. This will greatly simplify the design of efficient real-time storage, retrieval, and transport mechanisms to achieve high resource utilization.

Considering these objectives, we designed our novel Multi-Threshold Flow Control (MTFC) technique. It distinguishes itself from previously proposed algorithms by incorporating the following: (1) a multi-threshold buffer model, (2) a consumption prediction component and (3) a modular rate change computation framework in which new consumption prediction algorithms and feedback delay estimation algorithms can easily be incorporated.

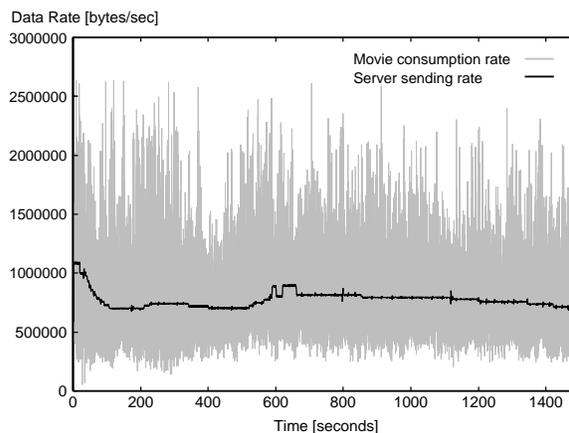


Figure 3: An example of an unsmoothed versus smoothed movie transmission profile.

The client playout buffer is a crucial component of any feedback control paradigm. The server is the data producer that places data into the buffer while the media decoder is the consumer that retrieves data. If the production and consumption rates are exactly the same then the amount of data in the buffer does not

change. If there is a mismatch, however, data will either accumulate or drain. If the buffer overflows or underflows then display disruptions will appear. Hence, the goal of managing the buffer is to keep the data level approximately at half the buffer size such that fluctuations in either direction can be absorbed. In an online feedback scheme, when the data level sufficiently deviates from the buffer midpoint, a correction message is sent to the server to adjust the sending rate.

The MTFC buffer manager can be configured with a variable number of data level *water marks* or *thresholds*. For example, the buffer space may be marked with 5, 9, or 17 thresholds (creating 6, 10, or 18 logical data partitions). If the buffer data level crosses one of the thresholds, then a speed-up or slow-down message is sent to the server. Note that the thresholds trigger a message only if the data increase or drain happens in the correct direction: thresholds above the midlevel send a slow-down message for data increases and thresholds below midlevel trigger a speed-up message when data continues to drain. The desired rate at which the server should speed up or slow down the transmission is calculated based on the current level of the buffer, its rate of increase or decrease and the predicted future transmission. It is designed to provide soft and infrequent rate changes during normal operation while taking corrective action aggressively when a buffer overflow or underflow looms. Note also that MTFC has a number of different threshold spacings at its disposal to aid in its operation. Thresholds may be spaced at an equal distance, geometrically, or logarithmically to provide different control characteristics.

From our extensive experiments conducted in a real-world environment we conclude that more than three buffer thresholds reduces the variability of the data transmission and the feedback overhead. At the same time, a consumption rate prediction algorithm allows the smoothing of streams with no prior knowledge of their transmission schedule. Therefore, our technique is very well suited for highly dynamic environments that need to adapt to changing network and load conditions. Furthermore, the achieved smoothing allows for improved resource utilization by significantly reducing peak data rates. The experimental results further show that our scheme outperforms existing algorithms in terms of traffic smoothness with

similar or less signaling frequency. Figure 3 shows an example of an original VBR movie transmission profile and the resulting schedule when using MTFC.

## 4 Rendering

The rendering side of the RMI system is composed of several parts. The video and audio streams are received over a sufficiently fast IP network connection on a personal computer running two instances of the Yima playback software. This media player is structured into several components and only one of them interfaces with the actual media decoder. This allows us to plug-in multiple software and hardware decoders and hence support various media types. For RMI, one of the players interfaces with a CineCast HD MPEG-2 decompression board manufactured by Vela Research. This decoder accepts MPEG-2 compressed high definition video at data rates in excess of 50 Mb/s and in both 1080i and 720p formats. An HD serial digital interface (SDI) connection transports the video to a high resolution front projector. Depending on the size of the venue, different projector models may be used. MPEG-2 video at a data rate of 40-50 Mb/s is referred to as having *contribution quality*. It is often used as the format of choice between production facilities and provides high visual quality. As a consequence, extending the visual field requires that the aural presentation be improved as well.

### 4.1 Immersive Audio

Audio can play a key role in creating a fully-immersive experience for a group of participants. To achieve this requires that the spatial limitations of traditional two channel stereo must be exceeded. Several rendering methods have been proposed that utilize digital filters to represent the spectral characteristics of sound sources from different directions in 3-D space (see [Kyr98] and references therein). These methods rely on accurate representation of Head-Related Transfer Functions (HRTFs) that represent the modifications imparted on sound by the head and the pinnae. To deliver sound over loudspeakers, these methods also require precise cancellation of the crosstalk signals resulting from the opposite side loudspeaker in order to deliver the desired sound

to each ear. As a result they work well only for a single listener in a precisely-defined position.

Multichannel surround sound systems have also been developed that use three front channels and two surround channels to provide a sense of envelopment for multiple listeners. Although these systems may work well in accompanying movies, they are not suitable for immersive systems. The main reason is that they leave significant spatial gaps in the azimuth plane (e.g. at 90 degrees to the side of the listeners) and provide no coverage in the median plane (no elevation cues).

In order to minimize localization errors the number of loudspeakers must increase linearly with the width of the listening area [RT90]. A listener that moves just a few cm from the designated listening spot is subjected to high imaging distortion and no longer experiences the correct localization cues. This problem can be addressed by increasing the number of loudspeakers.

We have designed and implemented a novel multichannel rendering system that addresses some of these limitations. It is known from the psychoacoustics literature that human listeners can localize sounds very precisely in the front hemisphere and less precisely to the sides and in the rear hemisphere [MM90]. Therefore, localization errors from the front channels that arise from offsets in the listener position from the desired center location will be particularly evident. In our implementation we allocate five channels to the front horizontal plane by augmenting the traditional three front loudspeakers with two additional wide channels. The advantages of this are twofold: (i) localization errors in the front hemisphere are reduced, and (ii) the wide channels provide simulated side-wall reflection cues that have been shown to increase the sense of spatial envelopment [And85]. In addition to the 5 front channels, a rear surround channel is added to fill the gap directly behind the listener. Elevation information is reproduced from two height channels placed above the front left and right loudspeakers. Early experiments that we have performed with this configuration have shown that these 10 channels significantly increase the sense of localization and envelopment for listeners.

The RMI implementation uses a second software player to interface with a 16/24-channel sound card model RME 9652 "Hammerfall." The audio data is

received via 16 channels in either 16 or 24 bit uncompressed linear PCM format at a data rate of up to 17 Mb/s. The audio is transported via ADAT lightpipes to a Digidesign ProTools system, and rendered via individual equalizers and powered speakers. Figure 4 shows the speaker locations of a typical 10.2 channel setup.

A major challenge in multi-listener environments arises from room acoustical modes, particularly in small rooms, that cause a significant variation in the responses measured at different listener locations. Responses measured only a few cm apart can vary by up to 15-20 dB at certain frequencies. This makes it difficult to equalize an audio system for multiple simultaneous listeners. Furthermore, it makes it impossible to render a remote performance for a large audience and have them experience the sound exactly as it is being produced at the remote event. Previous methods for room equalization have utilized multiple microphones and spatial averaging with equal weighting. This approach tends to smooth out large variations due to standing waves, but does not take into account the effects of room modes that for some frequencies may be more concentrated in one region and not in another. We have developed a new method that derives a representative room response from several room responses that share similar characteristics [BK02, BHK03]. This response can then be used to equalize the entire class of responses it represents. Our approach is based on a fuzzy unsupervised technique (c-means clustering) for finding similarities, clustering room responses, and determining their representatives. Our results show a significant improvement in equalization performance over single point equalization methods and we are currently implementing a real time version that can perform the necessary corrections on the incoming audio channels.

## 5 Applications and Demonstrations

The RMI system has been tested and showcased in a number of different configurations and locations. Yima servers were deployed locally in our laboratory on the USC campus and remotely at the Information Sciences Institute/East, Arlington, Virginia. Figure 2 illustrates the test environment used

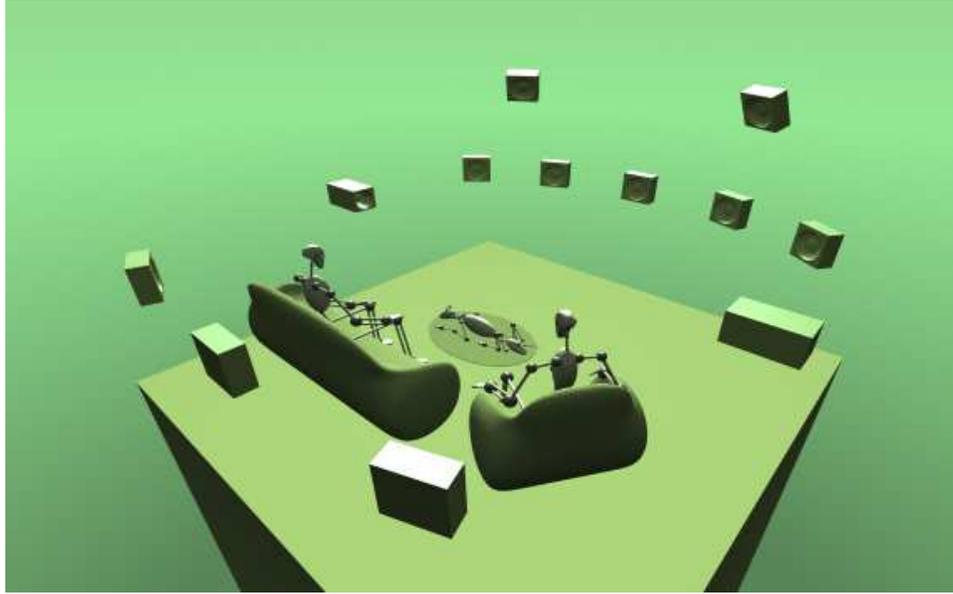


Figure 4: An immersive 10.2 channel audio setup illustrating the twelve speaker locations.

for some of our experiments. The connectivity was provided either through Internet2 or via a SuperNet link.

The client and rendering setup has been demonstrated at the following venues:

- The demonstration room next to our laboratory on the USC campus seats 10 people for an intimate, living-room type experience. This room contains a permanent setup and is our primary test location. Our first public demonstration of the RMI system took place here in June 2002. The equipment is continuously updated with the latest software and hardware versions and shown on a continuous basis to IMSC visitors.
- In October 2002 we presented a recorded performance by the the New World Symphony conducted by Alasdair Neale in USC's Bing Theater. This venue seats approximately 500 people and was specially equipped with a 30 by 17 foot screen and a 10.2 channel audio system for the RMI performance which was part of an Internet 2 Consortium meeting hosted by USC.
- In March 2003, the RMI system was demonstrated as part of the Internet2 Performance Production Workshop and the Internet2 Music Education Symposium at the New World

Symphony's Lincoln Theater in Miami Beach. Again, the venue was specifically equipped with a large screen and a 10.2 channel audio system.

RMI has shown itself to be adaptable to different situations. It also has proven reliable enough to be moved or recreated in different locations and for large audiences. Work to simplify the rendering setup (speaker calibrations, etc.) is currently underway at IMSC.

## 6 Discussion and Conclusions

The Remote Media Immersion system is the next step in audio-visual fidelity for streaming media delivered on-demand over the Internet. The goal of the RMI is to push the boundaries beyond what is currently available in any commercial system or other research prototype. Its emphasis is on the highest quality of audio-visual experiences and most realistic, immersive rendering. To achieve our goal we were faced with a number of unique challenges and had to design novel techniques to make RMI a reality. In this report we presented the error and flow control algorithms as well as the immersive audio aspects of RMI.

The current RMI setup is out of reach for most home users. For widespread adoption a number of technological advances will be necessary, which sub-

sequently will lead to more affordable prices and make the RMI system feasible for high-end home use. For example, we are currently using the MPEG-2 algorithm at a low compression ratio to achieve our target visual quality. An improvement in compression algorithms and affordable hardware availability will most certainly make the same quality available at lower bit rates in the future (MPEG-4 is a candidate here). Hence, we envision a cross-over point in the next few years, when the bandwidth required for RMI is below the bit rates that new high-speed, residential broadband technologies can deliver (for example very high-speed DSL, or VDSL). Additionally, we are working towards simplifying and automating the setup and calibration procedures that are currently necessary. Subsequently, these signal processing algorithms can be incorporated into multi-channel home receivers at moderate cost.

In the more distant future, one might envision a portable device with a high resolution screen and a personal immersive audio system. For a single listener it is possible to achieve very enveloping audio rendered with only two speakers, as long as the position and shape of the listener's ears are known. The PDA of the future will at some point have enough processing capabilities to combine visual tracking with adaptive rendering of both audio and video.

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