

## 摘 要

随着互联网与多媒体压缩等技术的快速发展，互联网流媒体应运而生。然而由于 Internet 与生俱来的尽力而为特性，以及在网络拓扑、传输介质和终端设备能力等方面存在的异构性，要求承载于其之上的流媒体系统必须能够提供一定的容错和网络自适应传输控制机制。由于可伸缩性是未来多媒体信源编码所追求的重要特性之一，本文主要针对可伸缩流媒体在互联网中进行传输时所碰到的若干与容错和适应性相关的关键问题进行深入研究。此外本文还研究了基于我国 AVS 标准的 P2P 网络流媒体系统实现，以及基于该系统的网络自适应丢帧算法。主要研究内容和创新点如下：

首先，针对单个非均匀容错保护框架在高码率情况下无法充分利用网络带宽资源的问题，提出了两种模块化非均匀容错保护模式和算法：基于码流分割和分组交织的非均匀容错保护模式，和基于字节交织和分组交织的双重交织非均匀容错保护模式。实验结果表明在不同的信源和信道条件下，这两种模式都能够取得接近最优的平均重构信号质量，而且后者还能够很好地保持可伸缩编码的渐进传输性能。

其次，研究了非均匀容错保护对于两种异构客户带宽多播情形的适应性问题。针对不同客户通过不同物理链路接入同一服务器的多播情形，提出了一个基于局部搜索的渐进优化算法，可降低为不同接入带宽分别计算最优容错保护解时的计算复杂度。针对不同客户局部共享同一物理链路接入同一服务器的多播情形，提出了一个基于嵌入式分组框架的非均匀容错保护机制及其性能评价函数，并以最优化该性能评价函数为目标提出了一个局部搜索算法和一个启发式算法。实验表明与现有定长分组分层多描述编码框架相比，所提出的嵌入式分组框架能够取得更好的客户平均性能。

再次，针对可伸缩视频非均匀容错保护和传输中的质量平滑问题，提出了一个基于加性上升/乘性下降的质量控制机制和一个网络自适应的在线比特分配算法。为了降低算法的计算复杂度，在采用基于局部搜索的渐进优化技术基础上提出了一个基于近似逼近的两阶段迭代技术用来加速算法的收敛过程。实验表明与均匀时间比特分配算法相比，该算法不仅能够以逐段平滑的方式在连续的帧组之间取得接近平稳或持续平滑递增的重构信号质量，而且具有较低的时间复杂度，适合于在实时流媒体系统中进行应用。

最后，设计并初步实现了一套 ISMA 兼容的 AVS 流媒体演示平台和一套基于 AVS 的 P2P 网络流媒体广播系统：AVStreamer。提出了两个基于优先级的网络自适应选择性丢帧算法，用于解决网络异构性所导致的 P2P 中继节点处 RTP 分组随机丢失问题。分析和实验结果表明即使在网络状况较差的情况下所提算法也能够取得连贯、流畅的播放效果。

总之，本文主要在深入分析现有流媒体传输技术不足的基础上进行研究，着重于通过应用层技术来改善流媒体传输的服务质量，提高系统的容错性能和对网络的自适应能力。所提技术已部分应用于实际系统并取得了良好的效果。

**关键词：** 互联网；流媒体；传输；可伸缩编码；信源信道联合编码；非均匀容错保护；质量平滑；比特分配；P2P 网络；AVS

# **Study on Key Techniques of Media Streaming over the Internet**

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With the rapid development of the Internet and multimedia compression technologies, it is possible to provide streaming media applications over the Internet. However, due to the best-effort property of the Internet and the heterogeneity of networks and terminals, it is necessary that the streaming media system must provide some capabilities of robustness and adaptiveness. Since scalability is one of the most important functions pursued by the next generation multimedia coding standards, the research of this paper puts its emphasis mainly on several key problems about the robust and adaptive streaming of scalable multimedia over the Internet. Moreover, this paper also studies the implementation of a peer-to-peer streaming media system which is based on the Chinese AVS standard, and two network adaptive frame-drop algorithms adopted in this system. The main research contents and contributions of this paper are listed as follows:

Firstly, in high bit rate conditions, if single unequal error protection framework is still used to protect and transmit a scalable multimedia bitstream, the available channel bandwidth resources can not be fully utilized, and the receiver can not obtain better picture qualities with the increased bandwidth. To address this problem, this paper proposes two grouped unequal error protection schemes: grouped unequal error protection by splitting with packet interleaving, and grouped unequal error protection by interleaving with packet interleaving. Experimental results show that both the two schemes can achieve near optimal average performance in various source and channel conditions, while the latter can still maintain the progressive transmission performance of scalable source coding in a very good extend.

Secondly, this paper studies the adaptiveness problems of multicasting unequal error protected scalable multimedia in two heterogeneous client bandwidth scenarios. In the scenario where different clients access the server via separate links, this paper proposes a local search based progressive optimizing algorithm, which can greatly reduce the computational complexity of searching optimal unequal error protection solutions for different client access bandwidths. In another scenario where many clients share a bottleneck link, this paper proposes an embedded packetization framework for layered unequal error protection, and a performance evaluation function for it. Aimed at optimizing this function, this paper also proposes a local search algorithm and a fast heuristic algorithm. Experimental results show that, compared with the fixed-length packetization framework in the literature, the proposed embedded packetization framework can achieve much better client-weighted

average performance.

Thirdly, to address the quality smoothing problem encountered while streaming unequal error protected scalable video over the Internet, this paper proposes an Additive Increase / Multiplicative Decrease quality control mechanism, and an online network adaptive bit allocation algorithm. To speed up this algorithm, a local search based progressive optimizing technique is adopted, and an approximate-approaching based dual-stage iteration technique is proposed. Experimental results show that, compared with the uniform-time bit allocation algorithm, the proposed techniques can achieve near constant or graceful increasing quality in a segment by segment scheme, and the accelerated algorithm is very efficient and can be used in real-time online applications.

Finally, this paper implements an AVS streaming media demonstration platform which complies with the ISMA (Internet streaming media alliance) specifications, and an AVS based peer-to-peer streaming media broadcast system: AVStreamer. Due to the heterogeneity of networks, there may be random packet-drops while relaying media data from upstream peers to downstream peers. To address this problem, two priority based network adaptive and selective frame-drop algorithms are proposed. Analytical and experimental results show that, even in poor network conditions, the proposed algorithms are able to achieve much fluent rendering effect.

As a conclusion, based on thorough analysis of some challenging problems in the area of media streaming, this paper tries its best to improve the quality of service via application layer techniques, so as to provide robust and adaptive media streaming of the Internet. The proposed techniques have partially applied to practical systems and achieved favorable effects.

**Keywords:** Internet; streaming media; transmission; scalable coding; joint source-channel coding; unequal error protection; quality smoothing; bit allocation; peer-to-peer networks; AVS