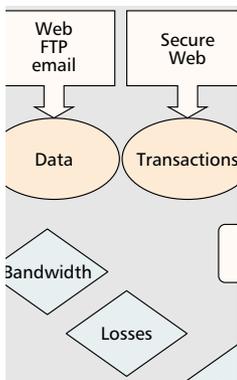


PROVIDING APPLICATION-LEVEL QoS IN 3G/4G WIRELESS SYSTEMS: A COMPREHENSIVE FRAMEWORK BASED ON MULTIRATE CDMA

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Wireless links provide only an unreliable communication channel that suffers from temporal outages. As a consequence, protocol mechanisms are needed that provide the different service qualities required by the emerging applications.

ABSTRACT

The emerging applications for 3G and 4G wireless systems typically require highly heterogeneous and time-varying quality of service from the underlying protocol layers. The wireless links, however, provide only an unreliable communication channel that suffers from temporal outages. As a consequence, protocol mechanisms are needed that, based on the unreliable wireless links, provide the different service qualities required by the emerging applications. In this article we identify the emerging IP-based applications for 3G and 4G wireless systems and categorize their QoS requirements. We discuss the wireless access mechanisms that show promise as the basis for supporting these applications. We then propose a set of protocol mechanisms that, based on the discussed wireless access mechanisms, provide the required QoS for the different application categories.

INTRODUCTION

The new degree of freedom wireless communication provides has made wireless devices extremely popular. According to a recent study there will be over one billion mobile phone users by 2003, and more people will access the Internet from wireless than wireline devices. The typical customer expects that all the communication services known from wireline communication will also be provided in the wireless format. Ultimately, customers want seamless communication services without any distinction between wireline and wireless systems. This, however, poses formidable challenges for communication service providers. First, in wireless communication, all the services asked for by the customer have to be provided over unreliable wireless links that suffer from frequent outages. Second, the wide variety of communication services (i.e., the appli-

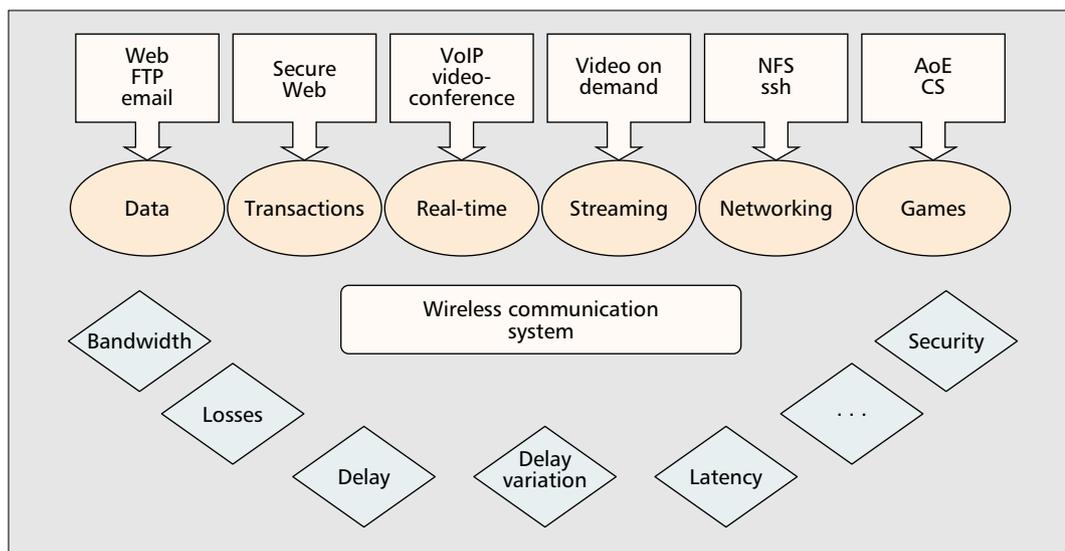
cations in network layer terminology) require vastly different quality of service (QoS) from the underlying networking protocol layers. Figure 1 gives an overview of the emerging applications for 3G/4G wireless systems and their QoS requirements. These applications and their various QoS parameters are discussed in detail in the next section. Each of these applications has different network traffic characteristics, which are also discussed. The different techniques with which these applications can be supported in a code-division multiple access (CDMA) system are discussed. The protocols that are promising candidates for supporting some of the emerging applications for 3G and 4G wireless systems are also covered.

APPLICATIONS AND THEIR QoS REQUIREMENTS

In this section we identify the emerging applications for 3G and 4G wireless systems. We characterize their traffic (i.e., bandwidth) requirements and discuss their delay and loss requirements. We propose a classification of the emerging applications based on their typical QoS requirements. In the IMT-2000 initiative — cdma2000, wideband CDMA (WCDMA) [1] — CDMA has been proposed as the wireless access technology for 3G wireless systems and beyond. Multirate CDMA in particular is a promising basis on which to support the emerging applications. This is because:

- The bandwidth requirements may differ vastly from one user to the other.
- A given user may require the support of variable bit rate traffic.

We discuss protocols that, based on multirate CDMA, provide the required application-level QoS for different emerging applications. We focus in particular on data applications (e.g.,



■ **Figure 1.** Emerging applications for 3G/4G wireless systems and their QoS requirements

Web requests) and audio/video applications (both for live feeds and streaming of prerecorded content).

Table 1 gives a systematic overview of the emerging applications for 3G and 4G wireless systems. Data applications consist of conventional data transfer such as Web transfers (using HTTP), file transfers (using FTP), and e-mail, as well as short message service (SMS). The traffic characteristics of the well-known applications Web file transfer and email have been studied extensively [2, 3]. Data applications do not tolerate any loss; they are, however, delay-tolerant.

A quickly emerging application for 3G and 4G wireless systems is games. In fact, Nokia recently unveiled a wireless gaming platform that is part of Nokia's Mobile Entertainment Services. The characteristics of gaming traffic depend on many factors, such as game design, game style (e.g., fast-paced "shoot 'em ups," or slow-paced strategic decision games), player experience, and playing style. The few studies

that have investigated gaming traffic (e.g., [4]) indicate that games have typically moderate bandwidth requirements on the order of 10 kb/s. The delay requirements are very tight for fast-paced games (typically on the order of 100 ms). The probability density function (pdf) of the packet interarrival times is a popular characterization of gaming traffic. Figure 2 gives the pdf of the packet interarrival times for the strategic game *Age of Empire (AoE)* measured in a typical wired LAN environment. Figure 3 gives the pdf of the packet interarrival times for the fast game *Counter Strike (CS)* measured for a multiplayer session on the Internet. (Both plots are obtained from measurements conducted in the Telecommunication Networks, TKN, institute at the Technical University Berlin).

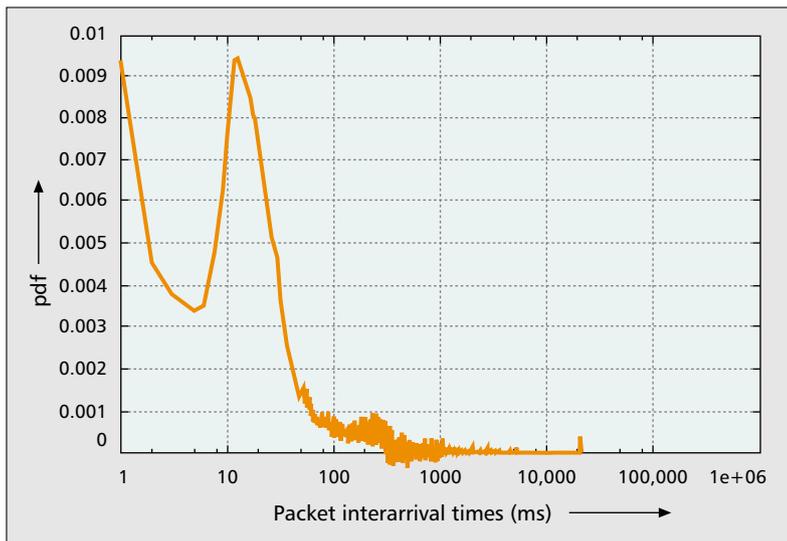
Other just emerging applications for wireless systems are network file system (NFS) applications. The drive toward wireless access to file systems is fueled by the availability of affordable high-powered laptop computers, which allow

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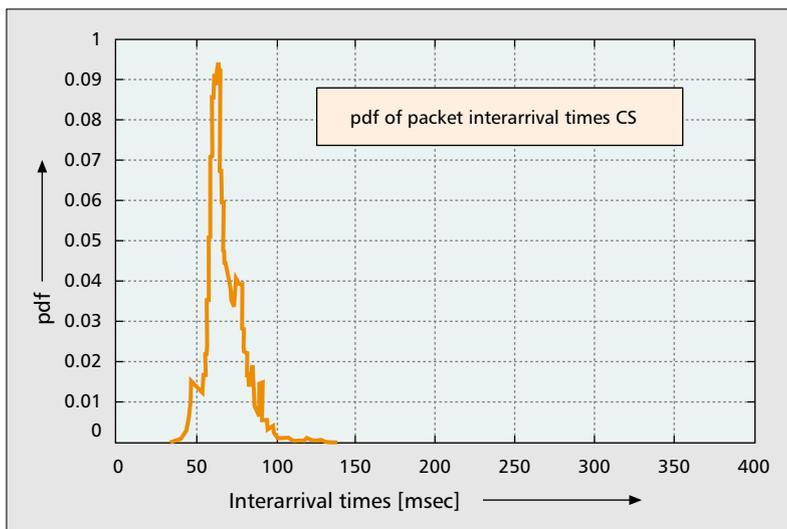
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Type of application and example			kb/s	Loss (%)	Delay (ms)
Data		FTP	Limitless	0	TCP timer
Real-time	Audio	PCM voice over IP	≤64	10 ⁻⁴	≤125 (with echo comp.)
		Voice over IP (codec)	10–64	10 ⁻²	≤125 (with echo.comp.)
	Video	MPEG-4	≤4000	10 ⁻²	≤125
		H.263	≤64	10 ⁻⁴	≤125
Non-real-time	Audio	CD	150	10 ⁻⁴	Playout buffer length
	Video	MPEG-4	≤2000	10 ⁻⁴	Playout buffer length
Games	MP	Strategic (AoE)	20	0–10 ⁻²	≤500
		Shoot+Run (CS)	20	0–10 ⁻²	≤100
Network service		NFS	Limitless	0	–

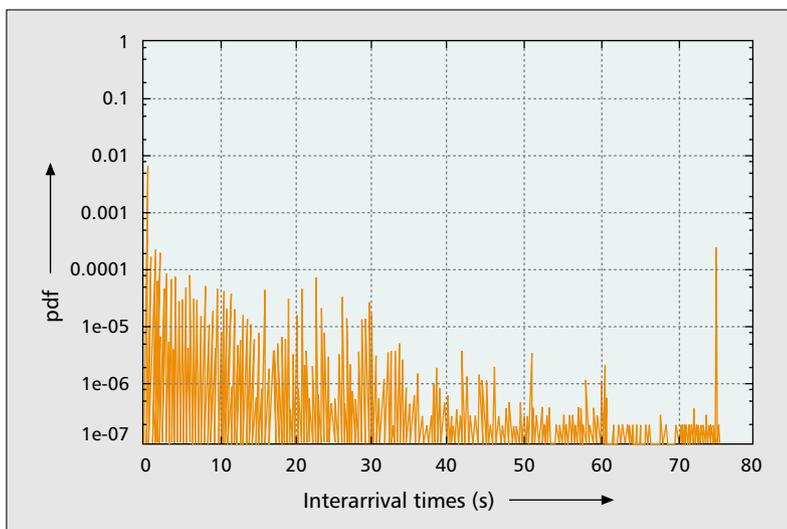
■ **Table 1.** Application requirements in terms of bandwidth, (one-way) delay, and loss for the different categories such as data, real-time traffic, non-real-time traffic, games, and network services.



■ Figure 2. Probability density function of packet interarrival time for the strategic game *Age of Empire (AoE)* over a typical wired LAN environment.



■ Figure 3. Probability density function of packet interarrival time for the fast-paced game *Counter Strike (CS)* over a wired wide-area Internet multiplayer session.



■ Figure 4. Probability density function of NFS packet interarrival time for a typical wired LAN scenario.

users to conduct their work anytime and anywhere. NFS traffic is expected to account for a sizeable portion of the traffic in 3G/4G wireless systems. It is therefore important to take the characteristics of NFS traffic into consideration when designing wireless traffic management mechanisms. However, NFS traffic has received little attention in the wireless community so far, and much more research is required. Figure 4 shows the pdf of the packet interarrival times of the NFS traffic on a wired LAN serving around 20 student workstations in the TKN institute at TU Berlin.

Real-time audio and video applications consist of conventional voice communication, voice over IP (VoIP), and videoconferencing. Moreover, audio and video feeds from live events such as conferences, concerts, and sporting events constitute real-time applications. Audio and video *streaming* applications, on the other hand, comprise the transmission of prerecorded audio and video content upon user request. In streaming applications the audio or video is played out at the user while the transmission is in progress. (In contrast, with file transfer, FTP, the entire audio or video file is downloaded before playback commences.) Figure 5 shows the pdf of packet interarrival times for the G.723.1 audio codec (VoIP) over a wireless LAN with no background traffic (obtained from measurements in the TKN institute at TU Berlin).

The traffic characteristics of MPEG-1 encoded video — which is suited primarily for wireline networks — have been studied extensively (see, e.g., [5]). The MPEG-4 and H.263 video encoding standards, on the other hand, have been specifically devised for wireless communication. These standards are expected to dominate in 3G and 4G wireless systems. The traffic characteristics of MPEG-4 and H.263 encoded video have only recently been investigated [6]. The traces have coefficients of variation in the range 0.3–0.9 and peak-to-mean ratios in the range 5–15. (The traffic characteristics of an MPEG-4 encoding of *Star Wars IV* is depicted in Fig. 6, see [6] for more details.) Moreover, the traces have long-range dependency properties. This indicates that the video streams are highly variable and bursty over many timescales. These findings have important implications for the design of video transport protocols in 3G and 4G wireless systems. First, allocating resources based on the peak rate of the video streams is highly wasteful. This is because video streams typically require the peak bandwidth only for short periods of time (usually during highly active scenes). Most of the time, the streams require significantly less bandwidth, as indicated by the large peak-to-mean ratios of the video traces. Efficient video transport can be achieved by statistically sharing the wireless transmission capacity among several ongoing streams. Packet-switched CDMA technology is very well suited to this packet-level multiplexing of several video streams. This multiplexing, however, requires the support of variable bit rates on the wireless links to the individual clients. Support for time-variable QoS is provided by different CDMA schemes, which we discuss next.

TIME VARYING QoS SUPPORT WITH CDMA

In this section we describe techniques that can be employed in CDMA systems to support variable bit rates on the individual wireless links. We discuss fixed spreading gain CDMA, variable spreading gain CDMA, and multicode CDMA techniques. We also discuss the importance of power control to support this variable rate traffic (Table 2).

FIXED SPREADING GAIN

In the fixed spreading gain approach [7], the spreading gain of each bitstream is kept constant. The chip duration is also kept constant. Users vary the transmission time and hence suffer from discontinuous transmissions. Low-rate users transmit for a smaller amount of time than do the higher-rate users. It can be seen that lower-rate users suffer from more multi-access interference than higher-rate users, since higher-rate users transmit for a larger amount of time. Higher-rate users, on the other hand, suffer from multi-access interference for a smaller amount of time. Hence, the performance for higher-rate users is better than that for lower-rate users. To compensate for this degradation in performance, lower-rate users transmit with more power to increase their signal-to-interference ratio.

VARIABLE SPREADING GAIN

The variable spread gain approach [8] relies on the assumption that the air interface bandwidth is fixed. Higher transmission bandwidths (bit rates) are achieved by reducing the spreading gain. The variable spreading gain approach is coupled with a variable power control that adjusts the energy per bit as the bit rate changes, thus maintaining a constant signal-to-noise and interference ratio (SNIR) across the range of bit rates. The variable spreading gain approach has increased signaling overhead since the receiver has to be informed about changes of the spreading gain. Another drawback of the variable spreading gain approach is high multipath interference (due to intersymbol interference, ISI) for short spreading sequences. Note that for a wireless link with a fixed delay spread the number of affected chips is constant; however, the number of affected symbols increases as the spreading gain decreases. Furthermore, high-power terminals (which typically use small spreading gains) may degrade the performance of low-power terminals [8].

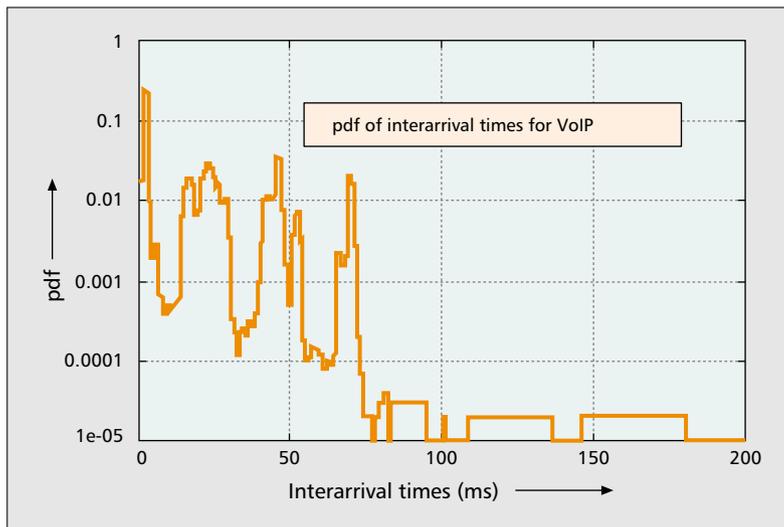


Figure 5. Probability density function of packet interarrival times for G.723.1 audio codec measured over wireless LAN with no background traffic.

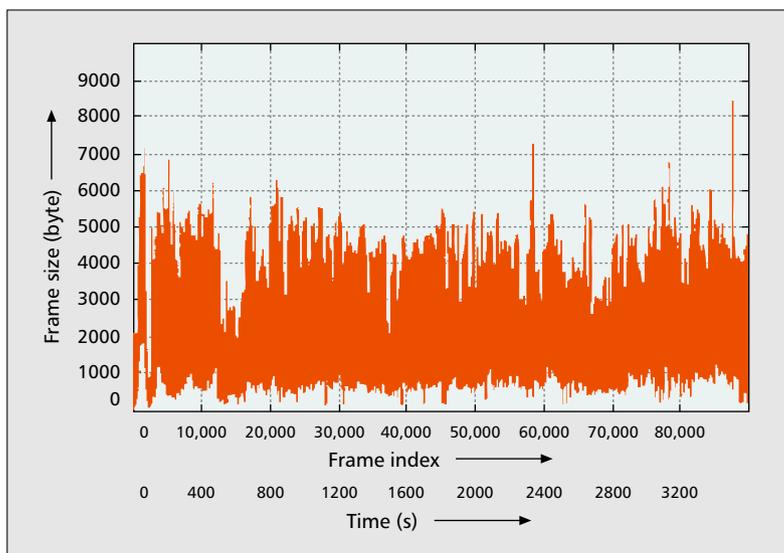


Figure 6. Frame size trace of Star Wars IV encoded using MPEG-4 at a high quality level.

MULTICODE CDMA

Multicode CDMA [9] can support a basic bit rate as well as integer multiples of the basic bit rate. In multicode CDMA a high data rate is split into smaller data rates. Each small data rate is then spread by a code sequence over the

	Fixed spreading gain	Variable spreading gain	Multicode CDMA
Multipath interference	High rate → high ISI	high rate → high ISI	No additional impact
Multiple access interference	Increases with bit rate	Increases with bit rate	Increases with number of channels
Hardware complexity	Synchronization	Additional oscillator	RAKE receiver for each channel
		High-performance power control	
Signaling overhead	Power control message	Power control message, change in spread gain	Not necessary
Granularity		High degree depends on spreading codes	Multiple of CDMA channels

Table 2. Benefits and drawbacks of multirate CDMA systems.

It has been noted that the multi-code approach is slightly more promising for multi-rate CDMA than the Variable Spreading Gain approach. It is argued that the former has a smaller signaling overhead and lower multipath interference.

entire coherent bandwidth. All spread signals are modulated and transmitted over the wireless link. The spreading codes have to be carefully selected to limit self-interference. One choice is orthogonal code sequences. These have no self-interference for synchronous transmissions, that is, downlink transmissions by a central base station or uplink transmissions on multiple codes by a given wireless terminal. However, the multipath effects may destroy the orthogonality on which the advantage of this scheme is based. For asynchronous transmission, orthogonal codes give poor performance. With orthogonal codes, the total number of code sequences in a given cell is limited. If a terminal wants to transmit on additional channels it has to request these channels from a central code depository. This introduces some latency for bit rate changes.

Another choice of codes is pseudo-noise code sequences. The number of these codes is large; however, the interference among code channels is high. Lin and Gitlin [9] proposed to use pseudo-random sequences to distinguish between the asynchronous uplink transmissions of the different wireless terminals. Additionally, they proposed to use orthogonal sequences to distinguish between synchronous uplink transmissions of a given wireless terminal. This approach gives a large total number of codes in conjunction with low self-interference between the parallel channels of a given wireless terminal. Further improvements are achieved when a central entity assigns the terminal-specific pseudo-random sequences [10]. Note that this does not require additional overhead for bit rate changes because the total number of codes is large and the codes are assigned to the terminals during link establishment. The hardware complexity, however, is higher since RAKE receivers are required for each channel to suppress multipath interference. It is noted in [8] that the multicode approach is slightly more promising for multirate CDMA than the variable spreading gain approach. It is argued that the former has a smaller signaling overhead and lower multipath interference.

PROTOCOLS FOR APPLICATION-LEVEL QoS

In this section we discuss protocols that are promising candidates for supporting some of the emerging applications for 3G and 4G wireless systems. These protocols are based on multi-rate CDMA. First, we discuss the Simultaneous MAC Packet Transmission (SMPT) protocol [11]. SMPT is designed to deliver transport-level segments (e.g., TCP or UDP segments, which are divided into several media access control, MAC, packets) with high probability within a prespecified delay bound. It achieves this by resorting to higher bit rates to recover from MAC packets lost due to errors on the wireless link. SMPT is a distributed protocol; it is designed primarily for uplink (terminal to base station) transmissions. Given these characteristics, SMPT is well suited to support uplink communication for applications with prespecified delay bounds and some loss tolerance. These are primarily games, as well as live audio and video streams.

Next, we discuss a protocol for the transmission of audio and video in the downlink (base station to terminal) direction, the prefetching protocol [12], which we refer to as W-JSQ. W-JSQ prefetches parts of the ongoing audio/video stream into buffers in the wireless terminals; it is thus primarily applicable to prerecorded content. With W-JSQ the playout of prerecorded audio/video at the wireless terminal may commence immediately upon user request (a short startup latency improves the protocol's performance). W-JSQ achieves small losses by prefetching portions of the ongoing audio/video streams, according to a join-shortest-queue (JSQ) policy. Channel probing is used to judiciously utilize the transmission capacities of the wireless links, which typically experience location-dependent, time-varying, and bursty errors. Roughly speaking, the base station schedules the packet for the terminal that has the smallest prefetched reserve and is currently experiencing good transmission conditions on its wireless link. The prefetched reserves help the terminal to continue playout during periods of adverse transmission conditions on the wireless links (and also when bursty high action scenes are played out). The simulation results in [12] demonstrate that JSQ prefetching in conjunction with channel probing is highly effective in reducing playback starvation at continuous media clients while achieving high bandwidth efficiency. For bursty VBR encoded video with an average rate of 64 kb/s and typical wireless communication conditions, the W-JSQ protocol achieves client starvation probabilities on the order of 10^{-4} and a bandwidth efficiency of 90 percent with client buffers of 128 kbytes. A variation of the W-JSQ protocol allows prefetching when transmitting continuous media in the uplink (wireless terminal to base station) direction. There is also an extension of the W-JSQ protocol that accommodates live audio/video feeds by introducing short playout latency (on the order of seconds), which allows the terminal to prefetch a short portion of the audio/video feed [12].

CONCLUSION

In this article we identify the emerging applications for 3G/4G wireless systems. We have given an overview of the QoS requirements and traffic characteristics of these emerging applications. We find that these applications typically require stringent assurances on delay and bandwidth. Moreover, they typically exhibit highly variable traffic (bit) rates. We argued that multirate CDMA is perfectly suited to accommodate these new applications. We outlined multirate CDMA techniques, as well as protocols that provide application layer QoS using these underlying multirate CDMA systems.

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BIOGRAPHIES

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