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Video Telephony Services over 3G Networks

Video telephony services over 3G networks are opening up a new range of possibilities for operators and service providers to allow users to communicate and interact with one another from any device, anywhere.

These services, while still in their infancy, require careful design and planning to match the user's high expectations for the 3G video experience. In particular, special attention needs to be paid to three key areas for effective service delivery – call setup time; synchronization and delay; and media quality.

Call Setup

Call setup is the time it takes before both video and voice start being transferred between two terminals. It involves the video call protocols running in the terminals and also the mobile network itself. It is critical to the success of mobile video telephony that users are able to talk and see one another almost immediately after the call is connected. Work is being done to design and specify enhancements to the established call connection protocols to enable communications in the minimum time possible, to the level where any delay is insignificant.

3G video calling is supported using circuit-switched bearers in the mobile network, using the 3G-324M protocol for the negotiation of the video session. The call setup time contributed by the mobile network is around 4s, and can be significantly longer for international calls, compared with around 2s for voice-only calls. Networks are now being optimized to equalize this difference in an attempt to make video call setup as close to that seen for voice services as possible.

Synchronization, Delay

Two aspects that affect a user's perception of mobile video telephony are delay and synchronization (lip-sync). Delay is a well understood challenge in all forms of telephony, especially the mobile variety, and voice services on mobile networks typically aim to have an end-to-end latency of not more than 150ms. Lip-sync is a new challenge which was only introduced when video was added; it is accepted that systems must maintain a constant lip-sync of within 50ms to enable natural face-to-face conversation.

With mobile video services the underlying network delays are sometimes as much as 300ms, although work is ongoing to optimize the access network to make this as close to the standard voice services as possible.

Maintaining good synchronization between audio and video channels is in direct contradiction to having low delays. It takes significantly longer to capture and encode a single frame of video for transmission than it does for the corresponding speech frames. This means the audio data must be delayed to match the video sequence in order to maintain synchronization between the streams, adding to the overall delay observed.

The problems are most evident when using multimedia gateways or other systems which rely on tandem transcoding, or decoding and recoding of the media stream. These types of gateways introduce lip-sync errors unless a compensatory delay is added. Newer types of gateways and systems which are able to convert between media formats more efficiently, without the need to decode and recode the media stream, do not produce significant lip-sync related delays.

Media Quality

Media quality is a function of the digital compression technique (Codec) used to encode the data for transmission. Video telephony services use a 64kbps bearer channel and the terminals must be optimized to make the best use of this limited bandwidth by using advanced digital compression techniques and efficient sharing of bandwidth between audio and video. Video bit rates are typically in the range of 45-55kbps, with a frame rate of 10-15 frames per second. Higher frame rates give better motion, but lower frame rates mean more bandwidth can be devoted to offering a sharper image, so this is an area where some balance should be made.

The digital chips used by the current generation of terminals are heavily loaded during video calls, which causes additional delay as well as limiting the quality of video seen by remote users. This further constrains the use of more advanced video compression standards, again reducing the quality of transmitted video.

As 3G terminals are enhanced with higher performance digital processors and are able to overcome the power limitations seen in the current generation, these quality constraints will no longer be a problem, allowing higher quality compression techniques to be adopted.

When an error occurs, there are error-checking techniques to allow the terminals to quickly restore the corrupted video image. This feature is built into all mobile terminals. But when calling with messaging servers and streaming applications, there is potential for having problems as these systems traditionally do not support advanced error-checking functions. Advanced video gateways, designed to operate in error-prone environments, available now work with the messaging platform more closely to overcome these issues and to maintain a higher quality service for

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the end user. These features however are not available in all currently available video gateways.

Circuit- vs Packet-Switched

Circuit-switched video telephony is currently the only viable way to deliver conversational video services, and this situation is likely to remain for some years. There are discussions on the future being with packet-based video services. However, they are unlikely to be able to offer the same quality as circuit-based services. There are several reasons for this: Although the available packet-switched bandwidth is 384kbps downstream and 64kbps upstream, these figures are not guaranteed, they depend upon how many other users are accessing the packet-switched services. Significant packet loss can occur in heavily loaded cells. In the packet-switched systems available now, there are insufficient quality of service (QoS) controls to maintain a video call.

When the IP overheads are applied the packet-switched connection only provides around 60% of the payload capacity that is available using circuit-switched based video telephony.

The 3G-324M circuit-switched protocols also incorporate a range of error protection techniques specifically designed to operate in mobile environments. They provide the facility for terminals to detect and recover from error-conditions in the live networks very efficiently, and are not available for IP data transport.

Development Direction

Until the packet-switched services can overcome these limitations, circuit-switched video services will continue to dominate through the use of optimized call connection times and improved quality video Codecs. Further enhancements are underway to allow circuit-switched video services to seamlessly inter-network with packet based applications to enable a "best of both worlds" solution.

The road to widespread adoption of video telephony is now wide and straight. The "maybe" application for 3G services a few years ago is now seen as a key element in the 3G portfolio offered to customers. Applications are being developed for consumer and business customers alike, and service providers are keen to take advantage of the benefits that face-to-face communications can bring in their interactions with customers.

Enhancements to mobile terminals and the underlying video telephony protocols and standards, to the mobile networks and to media gateways and multi-media systems are currently underway to assist with the onset of widespread mobile video telephony, to ensure that complete interoperability exists between the different services and to enable operators to bring services to market which will fully meet the expectation of their customers. ■

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