AT&T Developer Program

Guidelines for Delivering Streaming Services on AT&T’s Wireless Network

White Paper

Revision 1.0
Revision Date March 31, 2010
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Revision History

<table>
<thead>
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<th>Date</th>
<th>Revision</th>
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<tr>
<td>March 31, 2010</td>
<td>1.0</td>
<td>First release of this white paper.</td>
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1. Forward

This paper discusses considerations, recommendations, and guidelines to help developers efficiently deliver streaming media over the AT&T wireless network. The goal is to provide an optimal user experience for streaming services keeping in mind variables like device and wireless network characteristics.

1.1 Audience

The target audience for this white paper is software developers, architects, and managers who are considering development or deployment of streaming applications on the AT&T wireless network.

1.2 Contact Information

E-mail any comments or questions regarding this white paper via the AT&T Developer Program. Please reference the title of this paper in the message.

1.3 AT&T Resources

AT&T Developer Program: http://developer.att.com
Mobile Application Development: http://developer.att.com/mobiledevelopment
3G: http://developer.att.com/3G
Device Information: http://developer.att.com/devicedetails
Certified Application Catalog: http://developer.att.com/certifiedsolutionscatalog
Certified Application Catalog: http://developer.att.com/certifiedsolutionscatalog
AT&T Developer Program Resource on Platforms and Operating Systems

1.4 External Resources

3GPP Streaming Related Information: http://www.3gpp.org/ftp/Specs/html-info/26-series.htm
HTML5: http://dev.w3.org/html5/spec/Overview.html
Adobe Dynamic Streaming:  


Apple HTTP Live Streaming:  

Microsoft Smooth Streaming:  
1.5 Terms and Abbreviations

The following table defines the abbreviations used in this document.

**Table 1: Terms and Acronyms**

<table>
<thead>
<tr>
<th>Term or Abbreviation</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>3GPP</td>
<td>Third Generation Partnership Program</td>
</tr>
<tr>
<td>AAC-LC</td>
<td>Advanced Audio Coding Low Complexity</td>
</tr>
<tr>
<td>API</td>
<td>Application Programming Interface</td>
</tr>
<tr>
<td>AVC</td>
<td>Advanced Video Coding</td>
</tr>
<tr>
<td>CDN</td>
<td>Content Delivery Networks</td>
</tr>
<tr>
<td>CPU</td>
<td>Central Processing Unit</td>
</tr>
<tr>
<td>DRM</td>
<td>Digital Rights Management</td>
</tr>
<tr>
<td>EDGE</td>
<td>Enhanced Data Rates for GSM Evolution</td>
</tr>
<tr>
<td>EPC</td>
<td>Evolved Packet Core</td>
</tr>
<tr>
<td>EPS</td>
<td>Evolved Packet System</td>
</tr>
<tr>
<td>GOP</td>
<td>Group of Pictures</td>
</tr>
<tr>
<td>GSM</td>
<td>Global System for Mobile Communications</td>
</tr>
<tr>
<td>HE</td>
<td>High Efficiency</td>
</tr>
<tr>
<td>IDR</td>
<td>Instantaneous Decoding Refresh</td>
</tr>
<tr>
<td>HSPA</td>
<td>High Speed Packet Access</td>
</tr>
<tr>
<td>HTML</td>
<td>Hypertext Markup Language</td>
</tr>
<tr>
<td>HTTP</td>
<td>Hypertext Transfer Protocol</td>
</tr>
<tr>
<td>I-Frame</td>
<td>Intra-frame</td>
</tr>
<tr>
<td>IIS</td>
<td>Internet Information Services (Microsoft)</td>
</tr>
<tr>
<td>JPEG</td>
<td>Joint Photographic Experts Group</td>
</tr>
<tr>
<td>LTE</td>
<td>Long Term Evolution</td>
</tr>
<tr>
<td>MIME</td>
<td>Multipurpose Internet Mail Extensions</td>
</tr>
<tr>
<td>MPEG</td>
<td>Moving Pictures Expert Group</td>
</tr>
<tr>
<td>OS</td>
<td>Operating System</td>
</tr>
<tr>
<td>PCEF</td>
<td>Policy and Charging Enforcement Function</td>
</tr>
<tr>
<td>PCRF</td>
<td>Policy and Charging Rules Function</td>
</tr>
<tr>
<td>PSS</td>
<td>Packet-Switched Streaming Service (3GPP)</td>
</tr>
<tr>
<td>Acronym</td>
<td>Description</td>
</tr>
<tr>
<td>---------</td>
<td>-------------</td>
</tr>
<tr>
<td>RTCP</td>
<td>Real Time Control Protocol</td>
</tr>
<tr>
<td>RTP</td>
<td>Real Time Transfer Protocol</td>
</tr>
<tr>
<td>RTSP</td>
<td>Real Time Streaming Protocol</td>
</tr>
<tr>
<td>QCIF</td>
<td>Quarter Common Intermediate Format</td>
</tr>
<tr>
<td>QoE</td>
<td>Quality of Experience</td>
</tr>
<tr>
<td>QoS</td>
<td>Quality of Service</td>
</tr>
<tr>
<td>QVGA</td>
<td>Quarter Video Graphics Array</td>
</tr>
<tr>
<td>SoC</td>
<td>System on a Chip</td>
</tr>
<tr>
<td>TCP</td>
<td>Transmission Control Protocol</td>
</tr>
<tr>
<td>TS</td>
<td>Transport Stream</td>
</tr>
<tr>
<td>UDP</td>
<td>User Datagram Protocol</td>
</tr>
<tr>
<td>UMTS</td>
<td>Universal Mobile Telecommunications Service</td>
</tr>
<tr>
<td>VBR</td>
<td>Variable Bit Rate</td>
</tr>
<tr>
<td>VOD</td>
<td>Video on Demand</td>
</tr>
<tr>
<td>WAP</td>
<td>Wireless Application Protocol</td>
</tr>
<tr>
<td>WMV</td>
<td>Windows Media Video</td>
</tr>
</tbody>
</table>
2. Introduction

Audio and video streaming can work well over wireless networks, but need special considerations for delivery and optimization compared to wireline networks. The AT&T wireless network and devices offered by AT&T have extensive support for streaming. This paper discusses many of these capabilities, as well as techniques and guidelines that are likely to result in an optimal quality of experience given the unique characteristics of wireless network architectures, the wide range of available devices, and their particular characteristics.

Increasingly, wireless customers are using their devices, whether phones, netbooks or others, to access Internet services and data content. Of the many types of content available to users, audio and video media are especially sensitive to device capabilities and network conditions.

Video/audio delivery utilizes approaches such as downloads, streaming and progressive downloads. This paper focuses on streaming services since these are particularly vulnerable to wireless settings and conditions. From a user perspective, however, both progressive downloads and streaming can appear to provide similar experiences. Moreover, both approaches utilize similar technologies such as codecs and encoding tools. Therefore, many of the recommendations in this paper can be applied to both approaches.

Developing content and services that are delivered more efficiently is a win-win proposition for both developers and AT&T, because it can provide more users with a higher quality experience. It can also ensure that a wider range of devices can access streaming content. Ultimately, these guidelines benefit users by providing the best experience for their device.

Wireless in this paper refers to both cellular and Wi-Fi networks. Between these two options, Wi-Fi networks provide less restrictive media-delivery options.

The organization of this paper is as follows: key considerations for streaming over wireless; media delivery background on pertinent areas and variables that can affect user experience, AT&T key recommendations, and practices to guide developers in optimally using AT&T’s devices and network. There is also a short section on some anticipated capabilities. The appendix contains additional reference information.
3. Key Considerations

This section discusses some key considerations in developing media-delivery applications for AT&T’s wireless network.

Determining how best to stream content requires careful consideration of multiple factors. The user experience is affected not only by network conditions, but other factors such as the nature of the original content, the capabilities of the streaming protocol, and the characteristics of the user devices.

Examples of many important considerations include:

- The display resolution of different handheld devices, which can include sizes from 320X240 to 800X400.
- Available media players for different devices.
- Different codecs, their specific profiles, containers, and protocols that are supported on different devices.
- The processing capabilities of different devices.
- A wider range and varying throughputs and latencies on wireless networks compared to wireline networks.
- Effects of handover and mobility across different wireless technologies.

Table 2 below outlines the key elements that affect media delivery in mobile and wireless systems. Background on these elements is provided in Section 4. The recommendations and guidelines in Sections 5 and 6 are based on these considerations.

Table 2: Key Considerations: Layered View of Media Delivery Elements and Wireless Aspects

<table>
<thead>
<tr>
<th>Media Delivery Element</th>
<th>Explanation</th>
<th>Mobile/Wireless Aspects</th>
</tr>
</thead>
<tbody>
<tr>
<td>Content Type/Category</td>
<td>Content is either audio or video and varies by type of content and complexity such as level of details, activity, fidelity needs, etc.</td>
<td>Video content has to be shaped based on the content type, device screen size, resolution, and network throughput.</td>
</tr>
<tr>
<td>Media Delivery Element</td>
<td>Explanation</td>
<td>Mobile/Wireless Aspects</td>
</tr>
<tr>
<td>------------------------</td>
<td>-------------</td>
<td>-------------------------</td>
</tr>
<tr>
<td><strong>Content Delivery</strong></td>
<td>Covers communications method for delivering media information. Includes rate adaptations, protocols and delivery options such as real-time streaming versus progressive downloads.</td>
<td>There are several protocols that devices and the network support including 3GPP Packet-Switched Streaming Service (PSS) Streaming and simple HTTP delivery. Most handheld devices support mobile-specific protocols (e.g., PSS). Higher-end devices (e.g., smartphones, netbooks) may use the same protocols as other computing systems.</td>
</tr>
<tr>
<td><strong>Container</strong></td>
<td>File formats and other means of storing media information.</td>
<td>Feature phones use mobile-specific containers (e.g., 3GP). Higher-end devices may use the same containers as other computing systems.</td>
</tr>
<tr>
<td><strong>Content Encoding &amp; Codecs</strong></td>
<td>Algorithms for compressing and decompressing content. Many available codecs are available for audio and video (software- and hardware-based).</td>
<td>Mobile devices use the same codecs as other computing systems. Advanced codecs such as H.264 are well suited for wireless.</td>
</tr>
<tr>
<td><strong>Device Capabilities</strong></td>
<td>Various devices: lower-end handsets, smartphones, netbooks, notebooks</td>
<td>Capabilities vary across device types such as processing speed, resolution, and supported</td>
</tr>
<tr>
<td>Media Delivery Element</td>
<td>Explanation</td>
<td>Mobile/Wireless Aspects</td>
</tr>
<tr>
<td>------------------------</td>
<td>-------------</td>
<td>------------------------</td>
</tr>
<tr>
<td>Network</td>
<td>2G, 3G, Wi-Fi</td>
<td>Wireless networks have different characteristics than wireline such as bandwidth limitations, varying transmission quality, and mobility characteristics such as handover and roaming.</td>
</tr>
</tbody>
</table>
4. Media Delivery Background

This section provides background and details about the key media delivery elements for streaming, emphasizing those aspects that are unique to mobile and wireless use. The paper assumes that readers are already familiar with basic wireless and streaming concepts, including how to manage content and stream over wireline connections to desktop platforms.

4.1 Content Type

Audio Only Content
Audio streaming is well supported over AT&T’s wireless networks. The bit-rate required for producing a good quality audio stream is relatively low.

Video Content
There are two high-level categories of video content.
1) High-action content such as movies or sporting events.
When delivering action-oriented video, developers need to give additional attention to acquisition, encoding and delivery of the content.
2) Low-activity content such as people speaking, weather reports, and data charts.
Lower-activity content for mobile devices still requires careful attention, but can be delivered at lower bit rates.
A further categorization is between on-demand content versus live events. On-demand content is easier to develop and is the most common type of streaming application. One important consideration is to use the highest-quality source material possible.
Live-event streaming is more difficult to develop and requires more sophisticated infrastructure to deploy. Delivery protocols and encoding both require careful attention.

4.2 Content Delivery/Protocols: Downloads, Progressive Downloads, and Streaming

There are several ways to deliver media content to wireless devices. The three most common approaches are: download, progressive download, and streaming. For some AT&T Mobility devices, a form of streaming called simple Hypertext Transfer Protocol (HTTP) delivery is also possible, which can be adaptive or
simple (non-adaptive). The paragraphs below offer general background on each approach. Recommendations and guidelines related to the delivery methods are discussed in Sections 5 and 6.

4.2.1 Downloads

Downloads are the simplest way of providing media. A common example of a basic download is a podcast or a full-track song. The advantage is that the download can be accomplished with slower throughput connections. Another advantage is that from a developer perspective, this approach is the easiest to implement. Downloads are often accomplished via HTTP.

The primary downside is that the user cannot play the content until the download is complete. Additionally, the viewing device must have enough local storage to hold the entire media asset being downloaded. Finally, Digital Rights Management (DRM) needs to be considered when protected content is delivered using download. The DRM method depends on the license agreements with the content owners and the DRM methods supported by the devices.

4.2.2 Progressive Downloads

Though the user experience appears like streaming, progressive downloads are an extension of the simple download over HTTP approach described above. The key difference is that the content begins to play shortly after the download begins. This has the advantage of the user not having to wait for the download to complete. Some buffering is provided in case the download slows down. Progressive download is relatively easy to implement. Many of the downsides that apply to HTTP download also apply to this method of download. Progressive downloads are usually TCP-based, which is not the most efficient protocol, but is reliable. Popular sites like YouTube and Hulu utilize progressive downloads.

4.2.3 Streaming

Streaming refers to a continuous audio or video stream. An Internet radio station broadcasting its live feed online is an example of a streaming service. With streaming, the player presents the content almost immediately after receiving it (typically on the order of seconds), with minimal buffering.

Traditional streaming using Real Time Streaming Protocol/Real Time Transfer Protocol (RTSP/RTP) and HTTP-based adaptive, bit rate streaming can accommodate real-time changes in the delivered bit rate in response to network conditions.

Three primary approaches can be used to stream content:
a. 3GPP-Based, Packet-Switched Streaming Service

With this protocol, the content is delivered using RTSP/RTP protocols. The protocol has a built-in mechanism to adjust to network conditions and throughput availability, and to provide a continuous and smoother experience under varying networking conditions. This is accomplished by encoding audio and video at various bit rates.

This protocol requires specialized streaming servers to be able to optimally deliver the content.

b. Streaming over HTTP - Non Adaptive

Some new wireless devices support simple (non-adaptive) HTTP streaming delivery. The underlying technique is fundamentally the same as progressive download explained above with the difference being that the content is not cached in persistent storage, making it extremely difficult for the users to save the viewed content locally. This gives publishers greater control over their content.

Simple HTTP delivery avoids any visible effects of packet loss, because it uses Transmission Control Protocol (TCP) for transport. This ensures that any lost or damaged packets will be resent by the server.

This delivery method requires no dedicated streaming servers and can use an ordinary Web server. Though, since the server side usually consists of only a single-bit rate, non-adaptive file, simple HTTP delivery is subject to the playback stalling if the available throughput rate drops below the encoded rate for the content. Also, because the entire content object must exist on the server as a single file before clients can access it, this protocol is not suitable for live-event streaming.

c. Streaming over HTTP - Adaptive

HTTP Adaptive Streaming is explicitly supported by the iPhone using Apple’s HTTP Live Streaming protocol. Other server vendors are also moving towards supporting similar adaptive streaming technology. The key difference between the HTTP delivery described in the section above and this protocol is being able to accommodate real-time changes in throughput rates in response to network conditions. This is accomplished by encoding audio and video at several different bit rates.

This approach is also an improvement over most progressive download approaches, which risk experiencing stalled playback if network throughput drops below the media’s encoded bit rate for an extended period of time.
While Third Generation Partnership Program (3GPP) Packet Switched Streaming Service (PSS) streaming is primarily User Datagram Protocol (UDP-) based, most of the newer adaptive streaming technologies use the Web-standard HTTP protocol (which operates over TCP). The main benefits of using HTTP in streaming are to avoid connection failures relating to firewall restrictions and to obtain a reduction in bandwidth costs to the upstream servers, since HTTP adaptive stream chunks can be cached at an ISP’s Web caching servers or in content delivery networks (CDNs) at the edge of the network. There is active work going on (e.g., 3GPP PSS Release 9) to create a specification standard for HTTP adaptive streaming.

4.3 Containers

A “container” is a specification for storing audio, video, and/or other multimedia content in such a way that the synchronization of the audio and video is maintained and the content is organized for efficient playback. Containers support multiple media types including video, audio, text, and animation. Containers are also sometimes called “media containers,” “file formats,” and “wrappers.” When encryption or digital rights management technologies are used, these work in tandem with the container’s structure.

For feature phones, the most commonly used container is 3GP, defined in 3GPP specifications. This format is similar to the MP4 container, but is much more restrictive on what codecs can be inserted into the container.

Popular containers for higher-end devices such as smartphones or netbooks include FLV (proprietary to Adobe Flash), MOV (proprietary to Apple QuickTime), and WMV/WMA/ASF (proprietary to Microsoft Windows Media).

The TS (MPEG-2 Transport Stream) format is used in some streaming implementations such as Apple HTTP Live Streaming. (Note: an MPEG-2 Transport Stream does not necessarily contain video encoded with the MPEG-2 codec; the stream format and codec are independent of each other.)

Table 3 summarizes the codecs available for the different containers.

<table>
<thead>
<tr>
<th>Container</th>
<th>Video Codecs</th>
</tr>
</thead>
<tbody>
<tr>
<td>3GP</td>
<td>H.263, H.264, MPEG4 Part 2.</td>
</tr>
<tr>
<td>FLV (Flash)</td>
<td>H.264, On2 VP6, Sorenson Spark</td>
</tr>
<tr>
<td>F4V (Flash)</td>
<td>H.264</td>
</tr>
</tbody>
</table>
### 4.4 Content Encoding/Codecs

The term “codec” refers to the compression and decompression algorithms used to represent the video or audio. Video content uses one codec for the moving images and another codec for sound.

Codecs have a tradeoff between efficiency and computing complexity if the quality and resolution are kept constant. The newest codecs, which require more computations, result in fewer bits being needed to convey content with the same level of quality as older codecs with less aggressive processing requirements.

Encoding content to match a device’s capabilities is a critical part of preparing media content for mobile delivery. As a first step, the video and/or audio codecs must match what a device can support (for example, H.264 video and HE-AACv2 audio). Beyond that, the specific profile, level, and/or other constraints such as the device’s display resolution must be taken into account. (For instance, a target device might support video up to H.264 Baseline Profile at Level 1.2). The profile and level specify the capabilities of the device. Specifying Baseline, for example, constrains the encoding options to ensure that certain optional features of H.264 are not used, since using them would make the video impossible to play back on a target device. By specifying the Level to be 1.2, the combination of resolution and frame rate is constrained to be within the bounds of what the device can play.

One of the most popular and most efficient modern video codecs is H.264, also known as MPEG-4 Part 10 Advanced Video Coding (AVC), which was developed by the ITU-T Video Coding Experts Group together with the Moving Pictures Experts Group (MPEG). For desktop and laptop users, H.264 is currently supported by most modern streaming systems including Adobe Flash, Apple QuickTime, and Microsoft Silverlight. For newer devices, there is a good chance that H.264 is supported, regardless of the device’s operating system. While H.264 offers several different profiles of complexity, most handheld devices are limited to using the H.264 Baseline profile.

<table>
<thead>
<tr>
<th>Container</th>
<th>Video Codecs</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ogg (Open Format)</td>
<td>Theora</td>
</tr>
<tr>
<td>MP4</td>
<td>H.263, H.264, Dirac, VC-1, MPEG-4 Part 2 ASP</td>
</tr>
<tr>
<td>MKV</td>
<td>Sorenson, Cinepak, Real Video, Theora, H.263, H.264, Dirac</td>
</tr>
<tr>
<td>WMV</td>
<td>VC-1</td>
</tr>
</tbody>
</table>
4.5 Mobile Player Capabilities

Mobile phones from AT&T that support video and audio capabilities have media players. Media players can play media that is being streamed, as well as content stored on the phones. Many phones have built-in players. Third-party players are also available for some platforms.

4.6 Device Considerations

The device hardware plays a significant role in providing the desired experience for video or audio media delivery.

Thanks to faster processor speeds, more memory, and higher-resolution displays, the more powerful handheld devices are capable of supporting higher-resolution content. For the same codec, higher-resolution content translates directly to higher-throughput requirements.

Key device considerations, as shown in Table 4, are based on:

- **Hardware/Processor**
- **Screen Size and Resolution**
- **Codec and Protocol Support**

*Table 4: High-Level Device Considerations for Media Delivery*

<table>
<thead>
<tr>
<th>Type of Device</th>
<th>Media Capabilities</th>
</tr>
</thead>
<tbody>
<tr>
<td>Lower-end handsets (e.g., closed-operating-system feature phones)</td>
<td>Common audio and video codecs (e.g., H.264), but restricted selection. Mobile-specific containers (e.g., 3GP). Mobile-specific delivery protocols (e.g., 3GPP PSS).</td>
</tr>
<tr>
<td>Higher-end handsets (e.g., most feature-rich phones and most open-OS smart phones)</td>
<td>Common audio and video codecs. Wider choice than feature phones. Mobile-specific and common containers. Varies by platform. Mobile-specific and common delivery protocols. Wider choice than lower-end handsets.</td>
</tr>
<tr>
<td>Netbooks/notebooks</td>
<td>Greatest choice of built-in and installable players</td>
</tr>
</tbody>
</table>
Details on the video capabilities of devices offered by AT&T can be found at: [http://developer.att.com/developer/device_home.jsp](http://developer.att.com/developer/device_home.jsp).

### 4.7 Network Considerations

Most current devices offered by AT&T have 2G and 3G capability. Many support Wi-Fi. Throughput rates vary with the type of network, network conditions, and device capability. The key characteristics of the wireless network that can affect streaming services are: network type, mobility/handover, whether the user is roaming, latency, network utilization and congestion.

#### Network Type

2G is based on Enhanced Data Rates for Global System for Mobile Communications (GSM) Evolution (EDGE) technology and has the broadest coverage. 3G is based on Universal Mobile Telecommunications Service (UMTS) and High Speed Packet Access (HSPA) technology and is available in over 360 markets and to over 75% of the US population. Wi-Fi service is provided by hotspots or via private networks.

Typical download speeds for Edge networks range from 75 to 135 kbps whereas 3G (UMTS/HSPA) can range from 200 kbps to 3 Mbps, although AT&T’s low-end typical 3G speed is significantly faster. Upload speeds tend to be lower. Hotspot speeds vary based on the backhaul bandwidth and congestion. Actual throughput speeds experienced depend on multiple factors such as radio-signal quality, network loading and devices.

#### Mobility

Mobility is the key advantage of wireless services. But the speeds at which the user is traveling while accessing the network, as well as handovers between cell sites and between 2G and 3G systems, must be considered. Intra-system handovers are fairly seamless. The handover between 2G and 3G is also seamless from a networking perspective and allows the device to keep using the same IP address. It is important to keep in mind that a change in network type can result in different throughput capabilities.

<table>
<thead>
<tr>
<th>Type of Device</th>
<th>Media Capabilities</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>and support of codecs, containers, and delivery protocols.</td>
</tr>
</tbody>
</table>
Today, the AT&T network does not support handovers to or from non-3GPP networks including Wi-Fi networks. The device, in most cases, either maintains two connections (e.g., Wi-Fi and 3G simultaneously) or must establish a new connection. In both cases, the network does not assist in the handover.

**Roaming**

AT&T has agreements with domestic and international wireless carriers to give users a wider and more seamless coverage domestically, as well as internationally. Users have the ability to control or restrict data sessions when roaming internationally, as data sessions are generally billed at a higher rate, often on a usage basis, as opposed to the flat rate the user may enjoy on their home network. The ability to roam is a great asset for the users, but the experience may vary based on roaming agreements, as well as the capabilities and conditions of the visiting network.

**Latency**

Due to the complexity introduced by mobility, as well as device access protocols, wireless networks experience both higher average latency and more statistical variability in latency than wireline networks. A device’s media player buffer capability helps to mask the issues caused by delay, jitter, and handoff. Applications and players may need larger buffers in order to reduce the likelihood of the buffer running dry and creating a dropout. A bigger buffer may, however, result in a longer wait for the first video frame to be displayed. Refer to section 9.1 in the Appendix for further details on latency.

**Network Utilization and Congestion**

Streaming services must be designed for conditions created during high network utilization and congestion. Such conditions can exist both in wireline and wireless networks.
5. AT&T Key Recommendations for Media Delivery

This section presents AT&T recommendations for media delivery including type of networks to use, codecs, bit rates, and protocols. The next section provides additional guidelines for consideration to enhance the streaming experience.

5.1 AT&T Disclaimer

In addition to the disclaimers listed at the start of the paper, it is important to note that AT&T does not guarantee available bandwidth for sustained streaming sessions or persistent video or audio quality.

AT&T reserves the right to address bandwidth utilization, congestion, or quality. AT&T also reserves the right to specify performance characteristics.

AT&T wireless-data services may not be used in connection with high-bandwidth applications that disproportionately contribute to network congestion and function in a manner that is inconsistent with AT&T wireless-data services optimization requirements.

5.2 Selecting Network Type

Hotspots are part of AT&T’s wireless offering and should be used for multi-media streaming where feasible. For video streaming, it is recommended that applications use Wi-Fi as a default option in order to take advantage of AT&T’s Wi-Fi offering.

Media delivery can be done over any wireless network. More and more devices support Wi-Fi. Many devices will automatically connect to AT&T Wi-Fi hotspots.

Wi-Fi network selection can be done either under application control or user control.

If selecting Wi-Fi, developers should keep in mind that Wi-Fi traffic is not routed through the AT&T Wireless Application Protocol (WAP) Gateway, and developers will not be able to rely on WAP headers.

Refer to Table 10 in section 9.4 “Network Selection Interfaces” of the appendix for more details regarding methods and APIs for network selection.

5.3 AT&T Codec, Bit Rate, and Delivery Recommendations

This section presents AT&T recommendations on codecs, bit rates, and media delivery method.
5.3.1 Codecs

Where possible, content should be encoded in H.264 “Baseline Profile.” AT&T is working to ensure that more and more wireless devices support H.264 for video. Most devices offered by AT&T support H.264 “Baseline Profile” up to Level 1.2. A small number of low-end 3G devices only support level 1.0 and 1.1. Some of AT&T’s high-end devices support levels 2.2 and 3.0.

Some services support additional video codecs. For example, Windows Mobile devices use Windows Media player that can decode H.264, as well as Windows Media Video (WMV) content. Another popular codec is MPEG4 Part 2 that is most likely supported in higher-end mobile phones. For details on codec support per device, please refer to http://developer.att.com/developer/device_home.jsp.

5.3.2 Bit Rates

In a wireless environment, the target is an optimal bit rate. A higher rate is not always better, and can adversely affect the user experience depending on the network conditions and the device. When determining bit rates, data type, video resolution and frames rates supported by the device, as well as the access network, have to be taken into account.

As a general guideline, for video over 2G networks, bit rates between 44 kbps to 64 kbps help prevent excessive buffering. On 3G networks, bit rates between 85 kbps to 200 kbps can provide a good user experience. Wi-Fi can support higher bit rates.

As reiterated in the section below, the general guideline is to support multiple bit rate encodings in this range. Section 6.2.4 has more details on the frame rate recommendation.

All content should be encoded at multiple rates to allow optimum starting experience depending on the network type, as well as the capabilities and resolution of the device. Since the same content can be accessed by various devices and different network types and conditions, having this capability ensures the user can use a higher rate when and where it is available.

Content delivery should support adaptive delivery mechanisms wherever possible. See the section 9.2 “Rate Adaptation” of the appendix for background.

5.3.3 Delivery Method

AT&T recommends the 3GPP PSS Protocol be used for delivering the content in a continuous form of streaming. This recommendation is based
on the fact that most of the AT&T handheld device portfolio supports 3GPP Rel 6 PSS protocol.

This is the best protocol for supporting the widest range of devices and users.

This protocol uses Real Time Transfer Protocol (RTP) over User Datagram Protocol (UDP) for payload delivery. Since UDP has no inherent retry mechanism, the quality of the user experience can suffer during network congestion or handover to a slower network. The consequent visual artifacts can include buffering, pixilation, green screens, and distorted images. Hence, 3GPP Rel 6 Rate adaptation is recommended to dynamically adjust the content bit rate to the network conditions for delivering a smoother experience. Rate adaptation requires:

- Handsets that support 3GPP Rate Adaptation.
- Servers at the content providers or delivery point that support 3GPP Rate Adaptation.
- Content that is encoded using a multi-rate encoder. Note: these are available for both Video on Demand (VOD) and Live encoding.
- Tuning of the server/network pair for optimal support of rate adaptation.

Other streaming protocols such as simple HTTP delivery, HTTP adaptive streaming specific implementations such as HTTP Live Streaming and Microsoft Smooth Streaming can be used if supported by the device and the application.

Simple HTTP delivery is acceptable when it is supported by the target handset and the server can implement it according to AT&T guidelines. AT&T recommends that the developer design a solution to be implemented at the server that: (1) allows for the optimal delivery of traffic; and (2) implements mechanisms to minimize the bandwidth needed to deliver the content to the device. For example, the developer should limit the play-ahead time by maintaining, for example, no more than 10-20 seconds of content before playback. This recommendation also applies to progressive downloads. This minimizes unnecessary overloading of the network in cases in which a user starts watching a video using simple HTTP delivery, but then ends the playback session having only viewed a small segment of the content. Note that some HTTP servers support this solution on a per-session basis and Multipurpose Internet Mail Extensions- (MIME-) type basis (e.g., Microsoft IIS 7.0).
6. Additional AT&T Guidelines

The next section provides additional guidelines to consider for enhancing the streaming experience over the AT&T wireless network.

6.1 Content Source and Editing

Developers should select the highest quality source material that they can. If possible, they should use first-generation studio-quality original material in an uncompressed format. Each generation or compression step decreases the quality of sound and image quite radically.

If the content needs to be resized or edited in any way, developers should do the resizing or editing with the uncompressed source material using a high quality editing tool. Again, this will ensure the highest quality input into the encoding process that follows.

If uncompressed source format is not available, developers should use a source format that uses intra-frame- (I-Frame-) only compression, meaning content that only compresses information within the current frame. Examples include DV-25 and Motion-Joint Photographic Experts Group (-JPEG).

When resizing content, consider the target resolution for the range of devices that needs to be supported. If multiple resolutions and/or aspect ratios are required to support the range of desired devices, then independent edits and encodings should be done for each desired resolution/aspect ratio. It is recommended that the developer not rely on the handset to scale the video since this will degrade the overall end-user experience.

6.2 Encoding

6.2.1 Audio Only

Audio-only delivery requires lower throughput rates than video delivery.

For Audio, the codec of choice is HE-AACv2 if supported by the range of targeted handsets. HE-AACv2 is also known as aacPlus v2. Most handsets will decode HE-AACv2 even if they do not directly support the high efficiency (HE) features since the format is backward compatible with Advanced Audio Coding Low Complexity (AAC-LC). In this case, however, fidelity of the audio will be much lower. Note that handsets should be tested to verify that they can support HE-AACv2 either directly or in a backward-compatible manner since codec implementations do vary. The fallback is to use AAC-LC, which is universally supported by 3GPP Video capable handsets.
As a rule of thumb, the following general guidance applies for various content types using HE-AACv2 encoding:

- Low-quality mono (e.g., news/weather programs) – sampling rate 22 kHz, bit rate 24 kbps. Note: lower sampling rates may be acceptable for this type of content.
- Medium-quality stereo (e.g., sporting events and movies) – sampling rate 32 kHz, bit rate 32 kbps.
- High-quality stereo (e.g., music and movies) – sampling rate 44.1 kHz, bit rate 64 kbps.

6.2.2 Video Codec

Covered earlier in section 5.3.1.

6.2.3 Video Resolution

When resizing, developers should consider the resolution of the target devices and should encode separately for each desired resolution and aspect ratio.

The screen resolution of the handset typically influences the encoding resolution. For example, today a wide range of handsets have QVGA (320 x 240) displays. In the case of 4:3 content, it could be encoded at the full resolution of the display. For 16:9 content, however, the encoding resolution would be limited to 320 x 180 in order to avoid changing the aspect ratio or geometrically distorting the image.

When encoding, it is best to maintain the aspect ratio of the source content to prevent the introduction of geometric distortion into the encoded video.

6.2.4 Video Frame Rate

Video frame rate needs to be carefully selected based on the available bit rate, content type, and handset constraints. To maintain an evenness of motion, frame rates should match the source material if possible (up to 24 or 30 fps depending on the source). Lower rates may be necessary, however, either to maintain image quality at low bit rates or to maintain compatibility with a handset. When a reduction is needed, frame rates should either be reduced by a factor of two or three. In general, news or weather content is better suited to frame rate reduction than high-motion sports or music videos.

The majority of devices offered by AT&T are capable of video frame rates up to 15 frames per second (fps). Newer and latest devices are capable of frame rates up to 30 fps.
As a common principle, lower frame rates (e.g., 10, 12 fps) provide an acceptable user experience for news/weather programs and also use lower bit rates.

Highest frame rates are better suited for sporting events and music videos. Generally, 18 to 24 fps is sufficient even for high-resolution devices. A good rule of thumb is to use 15 fps or less for 2G/3G access and 24-30 fps for Wi-Fi access.

Over wireless networks, higher frame rates do not necessarily translate into a better customer experience.

### 6.2.5 Parameters

Using multi-pass encoding (also referred to as two-pass encoding) for on-demand (progressive download or download) content provides the best possible results at the desired bit rates. Developers should use adaptively tuned in-the-loop deblocking filters to smooth out blocking artifacts, resulting in a perceived smooth and clean image.

If the encoder allows a setting for the “Group of Pictures (GOP) Size” (i.e., the number of seconds between I-Frames or Instantaneous Decoding Refresh [IDR] frames [H.264]), setting it to a maximum of 2 to 3 seconds allows for smoother seeking within the video and also improves the visual experience in the event of packet loss. For live services, the setting required is likely closer to 1 second or less.

Developers should consider video encoders that have scene change detection capability and that set I-Frames or IDR frame accordingly. The normal I-frames/IDR frames interval setting should balance the following requirements: coding efficiency, refresh rate suitable to the network characteristics (e.g., packet loss, error rate), and seeking speed.

### 6.3 File Format

The encoded content file format should be 3GP using the 3GPP PSS-specified file format (i.e., an ISO MPEG-4 file format). In order to support both 3GPP PSS streaming and HTTP streaming with the same file the audio and video should be interleaved within the file and the metadata should be placed at the head of the file. This allows the file to be downloaded and played back concurrently.

For Apple Live HTTP streaming file formats, please refer to Apple developer information.
6.4 User Feedback

Where appropriate, developers should consider providing users information about the amount of data that content may consume. This may help users who are on usage-based data plans decide whether to access that particular content.
7. Streaming in the Future

A number of technology developments will enhance media-delivery applications in the future. One is radio-technology enhancements. Another is core-network enhancements, as well as intelligent gateways that will continue to use policies and enhanced capabilities. New Web standards such as Hypertext Markup Language (HTML) 5 will also provide additional options.

7.1 Radio Technology Advances

AT&T is continuing to upgrade its 3G UMTS/HSPA network to support higher data rates. This includes enhancements to support higher rates over the radio, as well as expanded backhaul capability. The result of both initiatives will be higher average throughputs and peak data rates, along with other radio management capabilities for improved services.

AT&T has deployed HSPA 7.2 technology across its 3G cell sites. As the standards for HSPA continue to develop, AT&T continues to evaluate and consider the technology and performance enhancements.

Beginning in 2011, AT&T is planning to deploy Long Term Evolution (LTE) technology that will enable even higher data rates compared to what is available with UMTS/HSPA. With LTE, users will be able to access a scalable IP network with “always on” capabilities to support a wide range of IP-based multimedia services and applications. While the LTE technology can support high theoretical peak rates in a 20 MHz channel, AT&T will begin deployments with narrower channels, supporting peak throughputs that are less than theoretical values but faster than today’s 3G technologies. As always, the device capabilities and its location relative to the wireless network, will also affect the data rates that are experienced.

7.2 EPC and Policy Enabled Networks

The policy framework, which gives networks more options over managing its data network, was introduced in 3GPP R7.

LTE networks will use an Evolved Packet Core (EPC) with a flat architecture that allows for optimal routing of media traffic. The EPC, which can also support 3G access networks and will evolve to provide a common UMTS/LTE core, continues to use the policy framework introduced in the earlier releases of 3GPP. The enhanced data handling capabilities of LTE together with intelligent Policy Elements in the EPC, i.e., the Policy and Charging Rules Function (PCRF) and Policy and Charging Enforcement Function (PCEF), comes with additional
enhancements for offering priority and policy for streaming applications. Figure 1 shows the Evolved Packet System (EPS) architecture.

**Figure 1: 3GPP Evolved Packet System**

![Evolved Packet System Diagram](image)

### 7.3 HTML 5

HTML 5 implements audio and video in web pages using the `<audio>` and `<video>` tags. This means that a separate media player (usually implemented as a plugin) is not required. At this time, the standard does not specify specific codecs or containers.

Opera, Chrome, and Safari support H.264 while Opera, Chrome, and Firefox support Ogg Theora. Currently, the only mobile browser that supports the HTML5 video element is the iPhone Safari.

The HTML 5 object model promises to allow full control via JavaScript, potentially allowing for JavaScript implementations of advanced features such as adaptive streaming.

While HTML 5 holds great promise, it is not yet final, and implementations are immature. As for netbooks and notebooks, Table 5 shows support for HTML 5 in popular browsers.
Table 5: HTML 5 Video Element Support in Popular (Non-Mobile) Browsers

<table>
<thead>
<tr>
<th>Browser</th>
<th>Theora</th>
<th>H.264</th>
<th>MPEG-4 Part 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Firefox 3.5+</td>
<td>Yes</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Chrome 3.0.182+</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
</tr>
<tr>
<td>Opera 9.52+</td>
<td>Yes</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Safari 3.1+ (on Mac OS X)</td>
<td>Yes</td>
<td></td>
<td>Yes</td>
</tr>
</tbody>
</table>
8. Conclusion

Providing applications over AT&T’s wireless network requires coordination between the developer community and AT&T. The objective of this relationship is to provide an optimal user experience and to support the growing numbers of devices, users, and increasing demand for multi-media services.

AT&T encourages the developer community to build and develop versatile applications that utilize various media. These applications not only need to be more efficient for wireless networks, but device capabilities and network conditions must also be taken into account. A key consideration is that more and higher is not always better when designing content for wireless networks. For example, higher encoding rates do not necessarily translate to a better user experience. As discussed in this paper, effective media delivery requires careful consideration at multiple levels including content creation, delivery protocols, codecs, device capabilities, and the specific networks used.

There are many exciting advances in progress that will continue to enhance multi-media experience for wireless customers. AT&T will keep bringing enhancements to both its network and its devices to make it the most effective network for the delivery of multi-media content and applications.
9. Appendix: Reference Information

This appendix provides streaming reference information that may be of interest to developers.

9.1 Latency

In addition to throughput, latency is another factor that can impact wireless media-delivery applications. Figure 2 shows the latency of different 3GPP technologies as measured from the user device to the edge of the network. Internet latencies are in addition to the values shown.

Figure 2: Latency of 3GPP Technologies

These are typical latencies for these technologies, and do not constitute guarantees for any particular network. There are many factors that determine the actual latency a user will experience.

Wireless network latencies have declined significantly with each successive technology. They are, however, still higher than wireline networks where access-network latencies are usually only a few milliseconds. Wi-Fi access networks, due to the short and relatively simple radio link, have latency on par with wireline networks.

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9.2 Rate Adaptation

Rate adaptation can accommodate varying wireless throughput. There are various means of accomplishing this, but the basic concept is that the content is encoded using multiple bit rates. Based on network conditions, the stream can dynamically adjust the bit rate of the content.

Figure 3: Visual Representation of HTTP Rate Adaptation Based on Available Bandwidth

Good conditions

Media split into series of file chunks. 1 to 4 seconds typical amount.

Different bit rates used under different network conditions.

Poor conditions

Most devices offered by AT&T support bit rate adaptation according to 3GPP Rel 6. A simple rate adaptation technique in this standard is to just fall back to a lower bit rate video segment for the remainder of the playback during diminished network throughput. This can be achieved based on feedback messages from the client. Devices also support much more complex rate adaptation techniques in 3GPP Rel 6.

Other rate adaptation systems are Adobe Dynamic Streaming, Microsoft Smooth Streaming, Apple Adaptive Streaming, Widevine, and Move Networks. Microsoft Silverlight supports adaptive streaming through what it calls smooth streaming in conjunction with an Internet Information Services (IIS) plugin.

H.264, VC-1, and On2 VP-6 are all codecs that can be used with adaptive streaming.

Commercial implementations use many of the ideas that are in Real Time Control Protocol (RTCP). RTCP relies on QoS and stream speed information to provide server feedback. For custom application development, rate adaptation
must rely on feedback from the client using methods like RTCP. Implementations periodically report throughput measurements for a recent time period. Typically, adaptive streaming is implemented in a combination of player and server. Most content developers employ a commercial adaptive streaming solution and do not develop their own.

Table 6 summarizes some of the details associated with different rate adaptation methods.

**Table 6: Details on Different Rate Adaption Methods**

<table>
<thead>
<tr>
<th>Streaming Approach</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>3GPP Rel 6 Packet-switched Streaming Service (PSS)</td>
<td>Requires client support from device vendor.</td>
</tr>
<tr>
<td></td>
<td>Rate change decision made by the server.</td>
</tr>
<tr>
<td></td>
<td>Relies on client feedback that may be inaccurate or delayed.</td>
</tr>
<tr>
<td></td>
<td>Requires all video sources to be encoded as multi-rate video.</td>
</tr>
<tr>
<td></td>
<td>Has no packet loss recovery capabilities.</td>
</tr>
<tr>
<td></td>
<td>Does not fully use excess bandwidth when available (no caching).</td>
</tr>
<tr>
<td>Microsoft Smooth Streaming and Apple HTTP Live Streaming</td>
<td>Requires client support from device vendor or video player supplier.</td>
</tr>
<tr>
<td></td>
<td>Rate change decision made by the client with timely local information.</td>
</tr>
<tr>
<td></td>
<td>Requires all video sources to be encoded as multi-rate video.</td>
</tr>
<tr>
<td></td>
<td>Has retransmission capabilities (TCP based).</td>
</tr>
<tr>
<td></td>
<td>Can fully use excess bandwidth when available (caching).</td>
</tr>
</tbody>
</table>

Most adaptive streaming approaches are for real-time streaming. It is also possible to use a progressive-download approach, and to provide the user means for selecting different bit rates. If the user experiences insufficient
throughput, the user can then manually select a low resolution. The download can then proceed from that point at a new bit rate.

9.3 3GPP Quality of Experience Metrics

3GPP has defined a number of Quality of Experience (QoE) metrics that are useful in assessing the overall quality of the user’s experience. These include:

- **Corruption Duration** – the time period from the last good frame before the corruption to the next good frame or the end of the reporting period (whichever is sooner). A corrupted frame is either an entirely lost frame, or a media frame that has quality degradation and the decoded frame is not the same as in error-free decoding.

- **Rebuffering Duration** – the duration of any stall in playback time due to any involuntary event at the client side.

- **Initial Buffering Duration** – the time from receiving the first RTP packet until playing starts.

- **Successive Loss of RTP Packets** – indicates the number of RTP packets lost in succession per media flow.

- **Frame-rate Deviation** – indicates when the actual playback frame rate during a reporting period is deviated from a pre-defined value.

- **Jitter** -- happens when the absolute difference between the actual playback time and the expected playback time is larger than a pre-defined value, which is 100 milliseconds.

- **Content Switch Time** – has a significant impact on the quality of experience for the user. The content-switch-time metric is used to report the time that elapses between the initiation of the content switch by the user up to the time of reception of the first media packet from the new content or media stream.

- **Lip-sync (i.e., audio/video synchronization)** – how well lip movements in the video match or align with the voice in the audio

Any field testing done should attempt to measure these metrics to assess the QoE and provide inputs to the optimization process. The challenge in capturing these metrics is they either require the video player/server to be instrumented or that the test sessions get captured in detail on the server using a tool such as Wireshark.
9.4 Network Selection Interfaces

There are APIs and interfaces for querying and controlling the current network type. These are available for Java Platform Mobile Edition, as well as for the various smartphone platforms, as summarized in Table 7.

Table 7: Methods and API for Network Connection Control

<table>
<thead>
<tr>
<th>Platform</th>
<th>Method and API</th>
</tr>
</thead>
<tbody>
<tr>
<td>Android</td>
<td>Use android.net.ConnectivityManager class to query and control the active connection.</td>
</tr>
<tr>
<td>BlackBerry</td>
<td>Use standard javax packages such as javax.microedition.io.Connector and URL option of 'interface.'</td>
</tr>
<tr>
<td>iPhone</td>
<td>Use Objective C, the Reachability API, and the Sockets API to check for connection types and open connections.</td>
</tr>
<tr>
<td>Symbian</td>
<td>With C++, use the Connection Manager API, such as RConnectionManager. With Java ME, use interfaces in javax.microedition.io.</td>
</tr>
<tr>
<td>Windows Mobile</td>
<td>With .NET Development, use the OpenNETCF.net package in the Smart Device Framework. With C++, use the Connection Manager API in connmgr.h.</td>
</tr>
<tr>
<td>Windows Mobile 7</td>
<td>Use the new Mobile Broadband API. COM based API declared in mbnapi.h. Old APIs from Windows Mobile 5/6 are still available.</td>
</tr>
</tbody>
</table>

9.5 Protocol Considerations

Table 8 has a high-level comparison and lists the advantages and disadvantages of streaming with TCP and UDP.

Table 8: Pros and Cons of Different Transport Protocols

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Advantages</th>
<th>Disadvantages</th>
</tr>
</thead>
<tbody>
<tr>
<td>User Datagram Protocol (UDP)</td>
<td>Simple protocol. Client latency doesn't affect server.</td>
<td>Transport layer does not support retransmission and cannot ensure that packets were received.</td>
</tr>
<tr>
<td>Protocol</td>
<td>Advantages</td>
<td>Drawbacks</td>
</tr>
<tr>
<td>----------------------------------------</td>
<td>----------------------------------------------------------------------------</td>
<td>---------------------------------------------------------------------------</td>
</tr>
<tr>
<td>No possible backlog. Many proprietary protocols are implemented on top of UDP.</td>
<td>Application layer needs to handle reassembly. Application layer needs to handle drop-off. Robust implementations are more complex.</td>
<td></td>
</tr>
<tr>
<td>Transmission Control Protocol (TCP)</td>
<td>Simpler to use than UDP from an application perspective. Sequence and application logic for proper reassembly handled by TCP. Well-known and easily handled by most networks.</td>
<td>Severe delays can be caused on slow networks due to TCP resend algorithm. Requires larger buffering by the server since OS will maintain resend buffers until acknowledged by client. Application layer requires drop-off logic if client gets too far behind in the stream.</td>
</tr>
</tbody>
</table>