



# Video-over-IP Communications

## **SIP versus RTMP**

### Introduction:

Today, if you have to manage Video Communications over Internet (public, private, wifi, 3g, 4g...), you may think about how you can solve some very common public IP networks' barriers and issues like:

- Packet lost UDP (no priority)
- Firewalls security policies don't allow to open any ports
- Standard VOIP packet filtering by carriers or routers
- UDP package sequence disordering
- Low bandwidth (traffic drop down)
- Coverage changes (wireless)
- Cellular saturations (wireless)
- Connection cuts off

This SIP vs RTMP Comparison Report, will help you to learn a bit more about these two protocols:

### Definitions:

The **Session Initiation Protocol** (SIP) is an IETF-defined signaling protocol widely used for controlling communication sessions such as voice and video calls over Internet Protocol (IP). It can be used for creating, modifying and terminating two-party (unicast) or multiparty (multicast) sessions. This telecom protocol has been well designed to work over private IP Networks with QoS or with enough bandwidth to ensure a good real time transmission.

More information: [IETF RFC 3261 - SIP Specifications](#)

The **Real Time Messaging Protocol** (RTMP) was initially a proprietary protocol developed by Macromedia for streaming audio, video and data over the Internet, and not only for a Flash player. Macromedia is now owned by Adobe, which has released the specification of the protocol for public use. RTMP has been designed and optimized for real time multi-media processing over the web.

More information: [Adobe - RTMP Specifications](#)

SIP / RTMP Comparison Table:

Features	SIP	RTMP
<b>Protocol</b>	ETF Open Standard	Abobe's public protocol
<b>Video Codecs</b>	H263, H263+, H264	H264*, H263 Sorenzon
<b>Audio Codecs</b>	G711, G729, GSM,...	Speex, G711
<b>Display Size</b>	16CIF, 4CIF, QCIF, CIF	Any specific display size L x h 16CIF, 4CIF, QCIF, CIF, VGA, HD...
<b>Video Decoding</b>	No error control during the decompression process. Packets lost will produce pixel's frames errors.	RTMP shows only complete frames, there's no errors on frames thanks to packet error management and smart buffering.
<b>Audio / Video Priority</b>	Yes, but it's not possible to play audio without video.	Yes, video can be stopped to prioritize audio
<b>IP Bandwidth</b>	Fixed, all bandwidth alloced will be used during all the call	Variable, you can change dynamically the bandwidth during a call.
<b>Frame rate</b>	Fixed, by the codec selection. Can freeze the video in case of network's issues	Variable, frame rate is automatically changed to maintain an optimal video and audi.
<b>Codecs Negotiation</b>	Yes, mandatory	Not required
<b>Packet Management</b>	No	Yes
<b>IP Packet</b>	UDP	TCP over HTTP or SSL
<b>Ports</b>	5060-5061 10001-20000 for RTP required to manage a SIP call	1935 (default) Only one ports opened, and can select any another port if required too)
<b>NAT</b>	No, a STUN server required or others NAT transversal technologies.	Yes, it works over NAT
<b>ID / Password Registration</b>	Yes, no auto-register option	Yes, with keep-alive and auto-register

## Why RTMP is solving better Internet network conditions than SIP?

### **Packet lost UDP (no priority)**

RTMP is using TCP, UDP Packet lost doesn't affect it.  
SIP send audio/video contents thru UDP packets.

### **Firewalls policies don't allow to open any ports**

RTMP only opens the 1935 port, required for web services with Flash; but it can use any other one.  
SIP opens many ports ranges like the 5060-5061 and other ones 10001-20000 ports for each call.

### **Standard VOIP packet filtering**

RTMP is not using VOIP standard packets, VOIP filtering for SIP or H323 doesn't affect it.  
SIP is easy to disturb or stop, some public routers and carriers are setting SIP-UDP packet filters.

### **UDP package sequence disordering**

RTMP include a package management and buffering to avoid errors.  
SIP will decode without a frame error control.

### **Bandwidth variations (lowered)**

RTMP is able to modify the frame rate and prioritize audio over video.  
SIP set the codec for each call.

### **Coverage changes / Loss of channel saturation**

RTMP works better than SIP on wireless networks 3G/4G or WIFI.  
SIP can't avoid to hang up the call on short connection coverages changes.

### **Connection cuts off**

RTMP is able to auto-register or keep alive the connection during small cuts off.  
SIP has to register again with a protocol negotiation process.

## Conclusions:

Obviously, when you are calling thru Internet over wifi or 3g or any other public Internet access, you will never ensure what routers, carriers, bandwidth, coverage, QoS are available. That's why, it's very difficult to use properly a SIP phone or soft-phone outside a VPN or a local network. To avoid previous IP networks common barriers, we strongly recommend to use better our [Flash/RTMP Server Channel for Asterisk](#) to make voice or video calls over the Internet instead of SIP. Try it and you will discover a really amazing solution for real time communications thanks to its very specific and advanced features.



Remember, you can connect SIP trunks/devices to RTMP with our VXI\*/Asterisk platforms to extend these new capabilities to your current standard VoIP services. So, you could use both SIP and RTMP according to the IP Network conditions where you have to manage your video calls.

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I6NET Solutions and Technologies is an European company dedicated to research and development of telecommunications and Internet technology. We are advanced services experts in voice and video interactivity (IVR / IVVR) in line with the latest evolution in telephony. Find out more information about I6NET at [info@i6net.com](mailto:info@i6net.com) or visit us at [www.i6net.com](http://www.i6net.com)

### **Corporate office**

I6NET Solutions and Technologies, SL  
Calle Magallanes, 13 - 5º Izq  
28015 Madrid (Spain)