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Structure and Practices of the)	CG Docket No. 10-51
Video Relay Service Program)	
)	
Telecommunications Relay Services and)	
Speech-to-Speech Services for)	CG Docket No. 03-123
Individuals with Hearing and Speech)	
Disabilities)	
)	

**Comments of the Rehabilitation Engineering Research
Center on Telecommunications Access**

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I. Introduction

The Telecom RERC (RERC-TA) is a joint project of the Technology Access Program at Gallaudet University and the Trace Center at the University of Wisconsin-Madison. The RERC is funded by the U.S. Department of Education, National Institute on Disability and Rehabilitation Research, to carry out a program of research and development focused on technological solutions for universal access to telecommunications systems and products for people with disabilities.

II. TRS Broadband Pilot Program (TRSBPP)

In its proposals for the TRSBPP, the FCC asks about the nature of the Internet services to be supported¹. Our measurements indicate that 256 kBit/s minimum bandwidth is insufficient to support acceptable call quality with the videophones currently available in the VRS market. These findings are discussed in detail below.

A. 256 kBit/s Minimum Bandwidth is Insufficient in the Current VRS Market

Implied by those questions is the assumption that minimum bandwidths of 256 kBit/s are sufficient for video calls, based on self-reporting by VRS providers. However, this assumption is not borne out by the facts in the current VRS market. The RERC-TA performed extensive measurements of the behavior of current VRS-provides videophones and software under bandwidth limitations. Our finding is that **300 kBit/s up and down** actually available bandwidth is the absolute minimum that must be supported under the

¹ FNPRM 11-184, Appendix A, at 3.

current H.263 video and G.711 audio interoperability standards, under otherwise perfect network conditions. Moreover, **220 kBit/s up and down** actually available bandwidth is the corresponding absolute minimum that must be supported for calls among **current videophones from the same provider**, which typically use H.264 video and G.711 audio. The detailed analysis and report supporting these findings are provided in the appendix.

It is also important to note that these numbers only set a floor with respect to whether with current video phones sign language communication with concurrent audio is supported and intelligible. They do not establish whether consumers are comfortable with the resulting call quality, and whether they subjectively consider the resulting video quality to be acceptable. Preliminary feedback by consumer advocacy representatives suggest that the subjective impression of the video quality on standalone videophones and laptop computers at these minimum levels is poor, and that higher minimums would be needed to obtain subjectively acceptable video quality that meets consumers' expectations. However, this latter area requires further study before definite conclusions can be drawn.

Finally, we also note that effective adaptive rate control (Section IV.B.1) could lead to tighter and lower overall bounds on the required available bandwidth, subject to the quality constraints for effective sign language communication. However, if the TRSBPP is expected to go into effect before the transition to the new technical standards for VRS is complete, the minimum bounds above stand – in particular, to ensure interoperability among legacy VRS-provided videophones for the purpose of making point-to-point calls,

the absolute minimum is 300 kBit/s up and down under otherwise perfect network conditions.

B. Transmission Rate Cannot be Equated with Bandwidth

Many VRS-provided videophones and software offer network configuration settings to adjust the videophone to the currently available bandwidth. Typical numbers include 192kBit/s, 256 kBit/s, 384 kBit/s, and 512 kBit/s. In some existing videophones the lowest configurable minimum is 256 kBit/s, while other videophones allow lower settings. Frequently, the user interface for these settings suggests that these numbers correspond to the speed of the user's network connection. However, posing this configuration option in this particular manner is highly misleading. It also seems that the minimum configuration setting of 256 kBit/s offered by some videophones has contributed to the mistaken assumption that 256 kBit/s available bandwidth are sufficient for VRS calls.

In fact, these settings correspond to the **raw average data transmission rate, not the required bandwidth**, in all videophones that the RERC-TA has tested. The actual bandwidth required for each of these settings is higher for two reasons: First, the packet overhead is not included in the rate calculations performed by the videophones. Second, the rates are only that – an average –, while in reality there are peaks and valleys in the transmission rates (shown in the graph in the Appendix V.A).

Although these peaks and valleys can be smoothed out without packet loss via sufficiently large transmission buffers, such buffers introduce additional latency. In some cases under tightly constrained bandwidth, the latency rises to 400 ms and higher one-way, which is too much to guarantee a good call experience. As we noted in an earlier

filing, the maximum one-way latency should not exceed 500 ms (in line with ITU-T recommendations for voice communications²). Furthermore, due to the fact that video processing is resource-intensive and itself induces some latency, the maximum recommended latency induced by the network should not be higher than 250 ms³. The only way to operate within these constraints is to overprovision the bandwidth relative to the raw transmission rate. With current videophones, for a transmission rate of 192 kBit/s this means an available bandwidth of 220 kBit/s, and for a transmission rate of 256 kBit/s (the minimum offered by some VRS-provided videophones), this means an available bandwidth of 300 kBit/s or better.

Another implication of the above findings and argument is that transmission rate control settings are crucial to getting a good call experience, yet at the same time they are confusing and misleading to consumers. Rather than asking consumers (or even VRS equipment installers) to select the correct rate for their network connection, it makes much more sense to require that all VRS equipment must implement adaptive rate control; see also Section IV.B.1.

C. Other Internet Performance Considerations

The FCC asks about latency and jitter, and the suitability of satellite Internet access for VRS communications⁴. As mentioned above, the maximum allowed latency for a high quality call experience is 500 ms, of which no more than 250 ms can be induced by the network. Satellite Internet misses this target by a wide margin; for instance VSAT

² ITU-T Recommendation G.114 One-way Transmission Time (2003). Online: <http://www.itu.int/rec/T-REC-G.114-200305-I>

³ Telecommunication RERC Filing, Section III.B.4. CG Docket 10-51, April 1, 2011. Online: <http://apps.fcc.gov/ecfs/comment/view?id=6016375091>

⁴ FNPRM 11-184, Appendix A, at 3.

Systems mentions that the best latency that it can achieve is 600-700 ms⁵. Consequently, the RERC-TA does not recommend satellite Internet access as a way to provide VRS to underserved areas, and urges the FCC to explore and exhaust all other means of providing broadband Internet access under the TRSBPP first.

Jitter is closely related to latency in that it requires equipment to set up jitter buffers, which hide the effect of packets that arrive out of order. However, any such buffers increase the latency, and thus jitter and latency need to be considered in combination – the criterion is whether combined jitter and network induced latency (due to transmission buffer sizes and packet transit times) still fall below the threshold of 250 ms.

III. Technical Implications of Per-User Compensation

In the proposal for structural reforms to the video relay service system, the FCC proposes a per-user compensation model, which implies that users will be locked into a single VRS provider for a certain period of time. Crucially, this proposal would also prohibit users from “dialing around” among VRS providers; that is, they would no longer be allowed to place a call through an alternate VRS provider⁶.

A. The FCC Did not Sufficiently Address Technical Reasons for Dial-Around

Some commonly given reasons for dial-around are related to the quality of service of VRS providers, such as the speed of answer and the quality of the interpreters. In a panel

⁵ VSAT “Latency – Why is it a big deal for satellite Internet.” Online: <http://www.vsat-systems.com/satellite-internet-explained/latency.html>

⁶ FNPRM 11-184, at 74-78.

meeting that involved representatives from the consumer groups, VRS providers, and the FCC, consumers brought up concerns with reaching 9-1-1 as another reason for allowing dial-around. The response by the FCC representative was that call quality and speed of answer standards for emergency calls would sufficiently address this concern⁷.

The RERC-TA contends that the question and the response miss the fact that there are also technical reasons for allowing dial-around. At any given time, the technical call quality across different providers can vary greatly, even when the user calls from the same location.

As we have noted in the past, “[f]irst, network congestion led to unusable video between the caller and VRS provider A. Switching equipment would not have helped, because the problem was with the network path between the caller and the VRS provider A. Moreover, dialing around to another provider with the same VRS app would not have helped, because the VRS app in question registers with the SIP server of the specific VRS provider A, which also was affected by the network problems. The only solution was to disconnect, close the VRS app A, and dial another VRS provider B with its own VRS app B, where at that time there were no problems with the network path between the caller and VRS provider B.⁸” In this scenario the video quality is not sufficient to maintain the conversation, and connecting via another provider resolves the problem.

⁷ Panel Discussion about VRS (FNPRM) Further Notice of Proposed Rule Making, Question by consumer representative: 13:00-14:12; Response by FCC representative: 17:45-18:30, Online: <http://vimeo.com/35601485>

⁸ Ex Parte filing on VRS Town Hall meeting by the Technology Access Program at Gallaudet University, 2/15/2012. Online: <http://apps.fcc.gov/ecfs/document/view?id=7021860437>

Crucially, problems of this type cannot be attributed only to problems with the users' and VRS providers' respective equipment and Internet connections. Call quality also depends on the characteristics of the network in between, something that neither the user, nor the VRS provider have any control over. If there is a degradation of the quality of service anywhere on the network path between the user and the VRS provider, the consumer suffers, and is not in a position to do anything about it, and neither is the VRS provider.

This situation is not limited to only emergency calls, but also can and does happen in everyday calls, including business calls. Removing dial-around capabilities would seriously hamper the users' ability to work around technical problems and actually push them further away from the goal of functional equivalence.

B. Choosing a VRS Provider is not Comparable to Choosing a VoIP or Wireless Carrier

The FCC contends that eliminating free dial-around would make VRS services more consistent with the way that most communication services are provided today⁹. Essentially, the argument is that people do not pick multiple VoIP providers and cell phone carriers, but rather engage in a contract with a single one. The RERC-TA strongly disagrees with this line of reasoning. Picking a VRS provider today is in no way comparable to picking a VoIP or wireless carrier.

Unlike VoIP providers and the services of wireless carriers, VRS providers are limited to offering their services on the open Internet, a network that they have no control

⁹ FNPRM 11-184, at 77.

over. They do not even have the option of engaging in service level agreements that make any kind of guarantees about quality and availability. This is in marked contrast to some VoIP providers, who route their calls over private IP networks, and thus control their quality of service to a much greater degree. For instance, Comcast notes that “there is a [...] type of VoIP [...], which in our case (Xfinity Voice) leverages our privately managed IP fiber optic network to deliver calls” and that “[t]he use of a privately managed IP network makes our phone service very different from the other types of VoIP. In addition to providing the reliability, safety and security all customers expect from their phone service.¹⁰” Note that Xfinity Voice is a consumer-level service, so these types of guarantees are not restricted to only enterprise users.

The principle of functional equivalence demands that deaf and hard of hearing consumers be able to use telecommunications in a similarly unrestricted manner as hearing people at similar cost. If hearing people run into problems affecting the call quality, they have the option of originating the call with a different provider; for instance through pre-paid phone plans, pay-as-you-go plans, adding a second contract, using access codes on PSTN, using a different interconnected VoIP provider, and so on. The costs for any of these options are nominal. Furthermore, they are available instantly, without requiring users to go through a waiting period to switch providers.

In contrast, if free dial-around were eliminated from the video relay service system and a technical problem arose beyond the user’s or VRS provider’s control, consumers would be left with two unappealing options: either to initiate a switch to a different VRS

¹⁰ <http://blog.comcast.com/2010/07/xfinity-voice-reinventing-whats-possible-with-your-home-phone.html>

provider and wait for the paperwork and the transfer to complete; or to bear the full brunt of the interpreting cost himself. Neither of these two scenarios is even remotely close to offering functional equivalence. It is bad enough that under the current system VRS users are unable to achieve full functional equivalence, due to the nature of the open Internet which makes call quality guarantees impossible, in contrast to what is available to the mainstream. Locking users into a single VRS provider would not serve to improve matters in this respect – quite on the contrary. In the mainstream world, this would be akin to eliminating all types of service level agreements from VoIP providers, and asking people to live with the consequences, no matter how bad call quality gets as a result.

At the root of the problem is that QoS for VRS calls is not supported end-to-end on the open Internet. If bandwidth and availability guarantees for the entire call path could be made, the situation would change. One possible way would be to integrate the provision of VRS with the provision of VoIP services (including integration with the IP Multimedia Subsystem). If the VoIP or wireless carrier also provides the VRS service and is in a position to guarantee call quality, these technical concerns would largely disappear – although it is still unclear how being locked into a single VRS provider is functionally equivalent to being locked into a single carrier, given the fact that paying for additional video interpreting is much more expensive than paying for an additional telecommunications carrier.

IV. Technical Issues

The RERC-TA applauds the FCC for taking the lead in ensuring that video relay services will be interoperable among one another, as well as with mainstream equipment.

Doing so is likely to raise the bar with respect to call quality for deaf and hard of hearing users, and also would open up the field to innovative ideas from research and development, both from within and outside the VRS industry.

It needs to be noted that following the standards cannot be optional, contrary to the wording of the question on whether specific protocols need to be mandated¹¹. For access to next-generation 9-1-1, the VRS service and videophones must be compatible with the NENA i3 Solution, so merely “encouraging” VRS providers to follow them is not sufficient – otherwise we would end up in the situation where users’ everyday calling equipment is incompatible with direct access to NG-9-1-1 services. In a similar vein, the VRS industry needs to track the activities of NENA and the EAAC closely to ensure that no hidden incompatibilities surface down the road.

A. Comments on iTRS Access Technology Standards

1. A New Interface between VRS Provider and Terminal Operator is Needed

The RERC-TA suggests adding an additional player to the diagram of the network relationships and VRS videophone interfaces¹², which we call the “terminal operator.” This entity constitutes the organization where the videophone is SIP-registered. Video Relay Service providers can provide this functionality, but they should not be the only ones to do so. In order to allow for true functional equivalence, the VRS users need also be allowed to be users of an application service provider for the SIP service, and associated communications, or else VRS users are put in the situation where they cannot

¹¹ FNPRM 11-184, at 46.

¹² FNPRM 11-184, Appendix B, at 19.

use their mainstream telecommunications equipment with the VoIP provider of their choice (or imposed by corporate policies).

This scenario applies to several important situations. First, it allows enterprise communication systems, which frequently run their own SIP server and set of terminals, to interoperate with VRS. Hence, the deaf and hard of hearing VRS users would get support for their calls and equipment just like mainstream users would. Otherwise, the VRS user would be unable to use the standardized equipment in the enterprise to call VRS, and have to rely on third-party equipment, which potentially runs into problems with corporate firewall policies, or is prohibited outright by corporate policies.

Second, it allows a natural path for interfacing session environments other than the open Internet with VRS. One key environment is the IP Multimedia Subsystem (IMS), which is a session environment that is expected to have importance for many users in the near future, particularly on wireless 4G services. The CSRIC III Workgroup 1 report notes that it is currently unclear how IMS will interface with relay services in NG-9-1-1 calls¹³: “Need to have specification developed to define how IMS interfaces with Relay Service.”

It is hard to see how videophones accessing IMS could be using VRS according to the model in Figure 1. IMS has SIP as its base, but it has some protocols on top of that of its own. It is likely that VRS users would like to use mainstream IMS Multimedia Telephony terminals registered by a 4G carrier for VRS access. The architecture in

¹³ CSRIC III Working Group 1 Final Report, December 2011, page 28. Online <http://transition.fcc.gov/bureaus/pshs/advisory/csrc3/CSRICWG1SG12ReportFINAL.pdf>

Figure 1 needs to allow for that scenario, which is best accomplished by introducing the terminal provider entity, which then also would provide the session border controller.

The fundamental tasks in establishing the iTRS access technology standards then consist of specifying the new interface between the terminal operator and the VRS provider, rather than only between the videophone and the VRS provider. The main function of the former would consist of providing an authenticated call interface for VRS calls. However, for interoperability on the open Internet, there also should be the option of the VRS provider running the SIP server, and allowing third-party off-the-shelf equipment to register with it.

2. General Capabilities

The list of functions in communication requirements section¹⁴ seems to miss the initial acquisition of the terminal location, usually by measurement if it is a mobile terminal, or by asking a LIS server if it is a fixed or wireless terminal. This could be added as point p.

The remote feature access mentions a visual incoming call alerting feature¹⁵. It is not sufficient just to have a visual flasher on the videophone, because it requires users to be in close proximity to their videophones. There needs to be a standard way to connect the incoming call alert to a housewide alerting system. Currently, this is accomplished either via an RJ-11 jack in the videophone, or by an incoming call detector connected to the router, which then connects to the housewide alerting system via an RJ-11 jack. However,

¹⁴ FNPRM 11-184, Appendix B, at 26.

¹⁵ FNPRM 11-184, Appendix B, at 27.

not all VRS providers offer this feature, and there is currently no established standard that ensures that this feature will work with all VRS providers.

In the user interface calling requirements¹⁶, the description of media is missing. The list of preferred media for communication needs to be established here, and made available as a choice of video, real-time text and audio, in any combination`. All three are important and should be supported simultaneously by terminals.

The call may result in a user-to-user call, a relayed call, or an emergency call with sign language support. The user-to-user call may contain any combination of the three supported media, e.g. real-time text - only, real-time text and voice, and Total Conversation = all three media. When the call is an emergency call, a series of extra actions needs to be performed, in accordance with the EAAC recommendations¹⁷ and the NENA i3 Solution¹⁸.

The user interface also must provide an easy way to toggle media on and off during a call. This is important for establishing video privacy, as well as muting the microphone input – echo and background noise picked up by the telecommunications equipment are a frequent source of annoyance in VCO and HCO calls.

¹⁶ FNPRM 11-184, Appendix B, at 29.

¹⁷ EAAC Report and Recommendations. Published January 26, 2012. Online: <http://www.fcc.gov/document/eaac-report-and-recommendations>

¹⁸ NENA 08-003. http://www.nena.org/?page=i3_Stage3

3. Specific Comments on the List of Standards

The RERC-TA applauds the compilation of the standards list for VRS access technology¹⁹, and how closely it aligns with mainstream standards, and the specifications for NG-9-1-1. The NG9-1-1 row should have a comment that location for mobile devices is usually acquired from measurements and network in some combination and not from a LIS server. Moreover, for emergency calls, the call should be placed simultaneously to the PSAP and a sign language interpreter (i.e., the sign language assistance, as specified by the EAAC recommendations P 2.2 and P 2.10). It also should be explicitly stated that for NG-9-1-1 calls all media formats supported by the PSAP and the terminal should be invoked in the media exchange phase.

In the same row, the draft phonebcp reference can be replaced by the published RFC 6443 standard.

4. Other Standards-Related Comments

The list of standards provided in Appendix B can only be a set of minimum requirements that VRS providers need to follow. To allow room for innovation, the VRS provider and the videophone must be able and allowed to support protocols and standards in addition to the ones listed as the minimum. These should not serve as an excuse for lock-in, however – today most VRS-provided videophones support SIP and H.264, yet in point-to-point calls across equipment from different vendors, they fall back to H.263, even though they are all capable of supporting better protocols and codecs.

¹⁹ FNPRM 11-184, Appendix B, at 32.

Finally, ensuring that VRS providers and videophones follow a common set of standards would open the playing field to third parties that are affiliated neither with telecommunication providers, nor with VRS providers. There also would be exciting potential for research groups to enter the field and to innovate functionality that specifically helps deaf and hard of hearing VRS users – one example of such functionality could be rate control suitable for sign language conversations, as discussed in Section B.1. Other possibilities include improvements for deaf-blind and low vision users, improved video compression technologies, and many others. In short, by opening up the VRS standards and allowing third-party videophones to connect to VRS, instead of having to rely only on VRS-provided equipment, we can expect to see rapid improvements in the usability of videophones for VRS calls. There also would be more incentive for VRS providers to partner with academia and other industry players, which would off-load some of the costs of developing videophone equipment from the TRS fund.

B. Other Technical Issues

1. Adaptive Rate Control

As we discussed in Section II.B, there are serious problems with setting the proper transmission rate of videophones manually. Aside from the potential confusion that it causes among VRS users, especially nontechnical ones, it also prevents videophones from reacting properly to changing network conditions. This problem is especially acute under severely bandwidth-limited Internet connections (such as the ones that are likely to prevail under the TRSBPP), and on mobile devices, where available bandwidth fluctuates greatly with the signal strength, and the presence of other mobile users in the same area.

The worst problems occur when the transmission rates are too high for the quality of the network connection. Even if it exceeds the available bandwidth only slightly, the result is substantial packet loss in the range of 10% and upward, as our measurement results in the appendix show. Transmission rates that are too low should not be ignored, either – as these result in lower call quality than what the users potentially would prefer, or outright unintelligible video.

The RERC-TA is pleased to see that RTCP is listed among the proposed standards for VRS technology²⁰. This is a good first step; however, getting rate control right in a manner that is compatible with sign language conversations is not trivial and will require substantial cooperative efforts. In particular, mainstream video calling software tends to make tradeoffs across frame rates, resolution, and image quality that are different from what is appropriate for sign language conversations. For example, we have observed that in the presence of network problems, or if the available bandwidth is low, Skype tends to reduce the frame rate below 10-15 fps, rather than reducing the resolution or the clarity of the image. This behavior is at odds with the minimum requirements for sign language conversations, which requires at least 20 fps for clear communication, or else users have to employ unnatural signing methods, such as slowing down, repeating signs, and asking for clarification^{21,22}. Rather than reducing the frame rate below acceptable levels, it

²⁰ FNPRM 11-184, Appendix A, at 32.

²¹ Harkins, J., Kozma-Spytek, L., Williams, N., Hellstrom, G., Vanderheiden, G., Ladner, R. (Jan 5, 2010). Ex Parte Comments of the Rehabilitation Engineering Research Center on Telecommunications Access and the MobileASL Project, In the Matter of Public Safety Issues Related to Broadband Communication To and From People with Disabilities, NBP #14, GN Docket Nos. 09-47, 09-51 and 09-137. Online: <http://fjallfoss.fcc.gov/ecfs/document/view?id=7020355298>

²² Telecommunication RERC Filing, Section III.B.2. CG Docket 10-51, April 1, 2011. Online: <http://apps.fcc.gov/ecfs/comment/view?id=6016375091>

makes more sense to reduce the resolution from CIF to QCIF, especially on mobile devices with small screens.

2. Camera Performance

The performance of cameras in low lighting conditions is an ongoing source of concern. While cameras in VRS-provided equipment and some tablets and laptops generally perform well under low lighting conditions, this state of affairs is far from universal. Users need to be able to conduct calls at dining room lighting levels (i.e. approximately 15 lux), and still be able to converse at 20 fps and upward. The maximum exposure time cannot be more than 40 ms at this level, or else motion blur prevents users from discerning important details in sign language conversations.

V. Appendix

This appendix describes the measurements that the RERC-TA took across videophones from three different vendors (Sorenson nTouch VP, Purple P3, and ZVRS Z4) under varying bandwidth limits and rate settings. All network captures and packet loss analyses were performed using Wireshark. Rate settings were adjusted in the videophone preferences. Simulations of limited bandwidth and associated buffer sizes were performed on a Linux-based Ethernet bridge running kernel version 3.0 (Ubuntu 11.10) using the Netem kernel module and the Token Bucket Filter.

A. Typical Rate Fluctuations during Calls

During a call, at a preset transmission rate, the actual data transmission rate is not constant. Rather, it fluctuates over time, with distinctive peaks and valleys, as shown in

Figure 1. The red and black line graphs show the fluctuations in the send and receive rates, respectively, with the x axis denoting the elapsed time in seconds, and the y axis denoting the used bandwidth in kBits per 100 ms. If the transmission rate is very close to the available bandwidth, the transmission buffers need to be sufficiently large to hold the peaks in the transmissions, in order to avoid packet loss, thus inducing additional latency into the call. Although the measurements shown in this figure were taken for a point to point call between two Sorenson nTouch videophones, they are highly characteristic of all types of video calls across all phones.

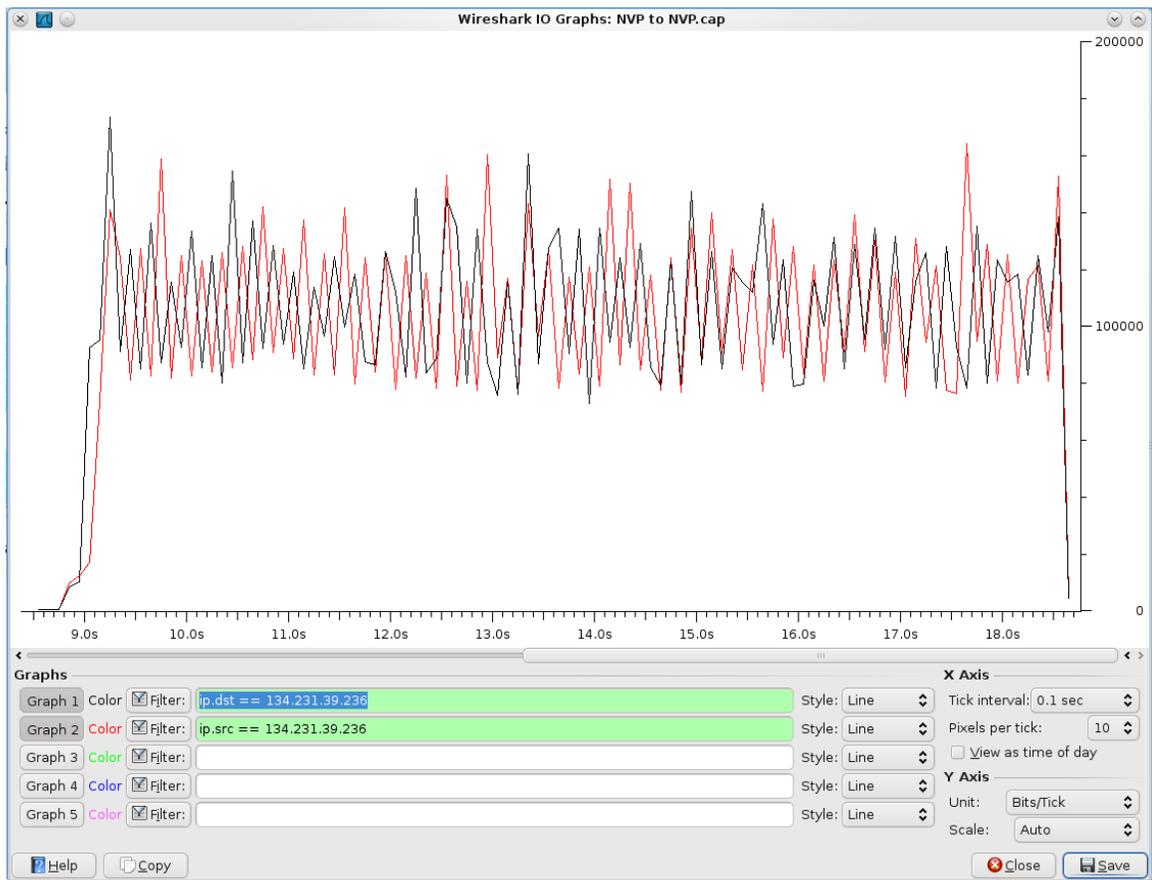


Figure 1: Characteristic rate fluctuations in video calls, demonstrating the need for overprovisioning bandwidth, or for latency-inducing buffering.

B. Problems with rate control in some cross-vendor video calls

Some videophones control their transmission rates better than others. If TRSBPP is to be implemented even before the transition to new VRS interoperability standards, bandwidth-limited users are likely to be exposed to it and run into severe interoperability problems as a result. One example of such rate control problems is shown in Figure 2, where the transmission rate was set to 256 kBit/s, but fluctuates wildly between 200 kBit/s and over 350 kBit/s. Fixing such rate control bugs in videophones is a prerequisite before minimum bandwidths in TRSBPP can even be considered.

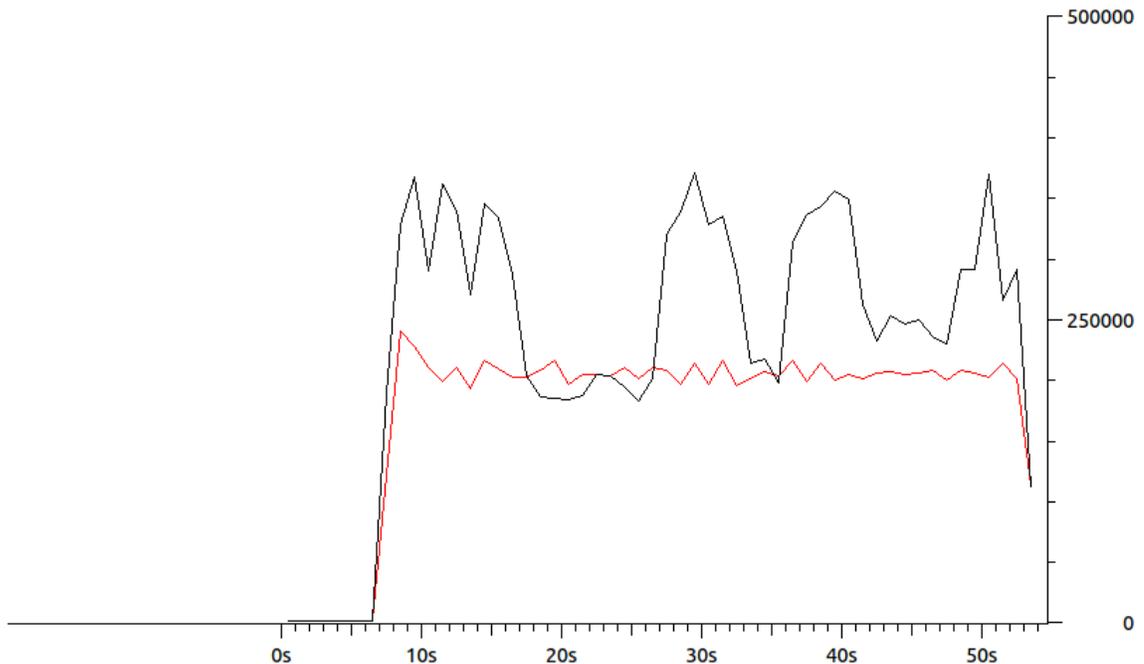


Figure 2: Rate control problems that can lead to interoperability problems if only the minimum bandwidth is available. The send rate (in black) fluctuates between 200 kBit/s and over 350 kBit/s, even though the transmission rate was set to 256 kBit/s. If a connection is limited to 300 kBit/s available bandwidth, severe packet loss would occur.

C. Minimum Bandwidth Measurements

The table below shows the measurements across the Sorenson nTouch VP (nVP), Purple P3, and ZVRS videophones, and demonstrates the tradeoffs between bandwidth, rate control, and latency.

Send VP	Send Rate in kBit/s	Recv VP	Recv Rate in kBit/s	Bandwidth in kBit/s	Buffer Latency in ms	Video packet loss %	Audio packet loss %
nVP	192	nVP	192	200	56	10.5	4.1
nVP	192	nVP	192	200	176	10.8	4.9
nVP	192	nVP	192	200	296	9.7	5.1
nVP	192	nVP	192	200	416	9	5.2
nVP	192	nVP	192	200	536	11.2	4.4
nVP	192	nVP	192	220	50	0	0
nVP	192	nVP	192	220	25	1.3	0.2
nVP	192	nVP	192	215	50	3.7	0.4
nVP	192	nVP	192	215	100	2.8	1.2
nVP	192	nVP	192	215	200	2.4	0.5
nVP	192	nVP	192	215	400	3.6	1.5
nVP	192	nVP	192	215	800	4.1	1.4
nVP	256	nVP	256	270	50	8	2.3
nVP	256	nVP	256	270	100	7	2.3
nVP	256	nVP	256	270	200	6.7	2.9
nVP	256	nVP	256	270	400	7.1	2.8
nVP	256	nVP	256	270	800	6.6	2.2
nVP	256	nVP	256	280	50	2.9	0.9
nVP	256	nVP	256	280	100	3	0.9
nVP	256	nVP	256	280	200	2.5	0.7
nVP	256	nVP	256	280	400	2.9	0.7
nVP	256	nVP	256	280	800	2.7	0.7

nVP	256	nVP	256	290	50	0	0
nVP	256	nVP	256	290	25	0.1	0.1
P3	256	nVP	256	290	50	0.4	0
nVP	256	P3	256	290	50	6.7	4
nVP	256	P3	256	290	100	4.3	2
nVP	256	P3	256	290	200	3.2	1.1
nVP	256	P3	256	290	400	6.4	1.7
nVP	256	P3	256	290	800	6.7	2.5
nVP	256	P3	256	300	50	7.9	4.6
nVP	256	P3	256	300	100	5.1	2.6
nVP	256	P3	256	300	200	6.8	2.8
nVP	256	P3	256	300	400	8.8	4.3
nVP	256	P3	256	300	800	2.3	1.2
nVP	256	P3	256	300	1600	0	0
Z4	256	P3	256	300	50	1.2	0.1
Z4	256	P3	256	300	100	0.9	0.2
Z4	256	P3	256	300	200	0.8	0.1
Z4	256	P3	256	300	400	0	0
Z4	256	P3	256	300	800	0.1	0
nVP	256	Z4	256	300	50	14.4	1.8
nVP	256	Z4	256	300	100	13.9	4.9
nVP	256	Z4	256	300	200	14.1	2.3
nVP	256	Z4	256	300	400	13.7	2.2
nVP	256	Z4	256	300	800	11.9	1.9
nVP	256	Z4	256	300	1600	0	0

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