Voice-over-Internet Protocols: a new dimension for translation interaction

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Abstract. *Voice-over-Internet Protocols (VoIPs) have become of great help to freelance translators in a short period of time, giving them the possibility to communicate with professional colleagues all over the world. They help solve problems by reducing the time needed for research and the cost of phone calls. This paper considers the technological nature and requirements of VoIPs, discussing the developments and technical limitations encountered over the last decade, as well as recent strategies to overcome difficulties. The reliable services of Internet telephony are possible only if two changes are produced: 1) the improvement and deployment of the IP/ATM/synchronous optical network and ISDN, cable modems and x digital subscriber line (xDSL) technologies, and 2) the sampling payment for public Internet. Unfortunately there are few objective criteria for assessing the efficiency and the applicability of these tools to the work of translators. Further study on the influence of Internet communication on translators and interpreters' work is needed.*

Introduction

The feeling of isolation and the inability to participate in real-time communicative exchanges while at work have always been downsides of working as a translator, especially in the case of freelancers. Although working in a company is not necessarily synonymous with a friendly and cooperative environment, we all feel the need to check translation solutions or seek some help with terminological doubts. The introduction of excessively hyped CAT technologies does not seem to have overcome these obstacles. Yes, they may have helped us improve efficiency, but the partial automation of the translation process probably makes human-to-human interaction even harder.

The appearance of Voice-over-Internet Protocols (VoIPs) has brought the reality of real-time interaction closer than ever before. Freelance translators are now able to communicate with professional colleagues and informants from all over the world. This is particularly important when we consider the time required to find a valid solution for the unpredictable number of doubts we may be faced with, while the economic advantages of drastically reducing our phone bills are obvious.

Apart from one-on-one interaction, which is usually the quickest way to tackle a particular problem, we should also envisage a constantly updated worldwide real-time translation forum via the Internet, where everybody could voice their opinion. In this way, the various netmeeting applications could provide dynamic new possibilities beyond writing.

The main aim of this article is to discuss the technological nature and requirements of VoIPs, whose sudden emergence in the late 1990s and rapid dissemination in the present decade have given rise to a legion of ardent supporters, and to more detractors than expected.

The possible applications to translation and distance teaching are multiple. However, many doubts have arisen about their quality and feasibility in language projects, as the increasingly heavy use of the Internet's limited bandwidth often results in congestion and delays in transmission.

Let us now briefly survey the history of Voice-over-Internet Protocols and their quick evolution as far as quality is concerned over the past decade.

The Evolution of VoIPs

The possibility of voice communications traveling over the Internet, rather than the typical and still predominant public switched telephone network (PSTN), first became a reality in February 1995, with the introduction of the first Internet Phone software by the company Vocaltec Inc. The software was designed to run on a 486/33-MHz PC (which now seems almost prehistory), equipped with a soundcard, speakers, microphone and modem. The software compressed the voice signal and translated it into IP packets for transmission over the Internet. The downside to this system was that both parties had to be using Internet Phone software, otherwise communication was impossible.

Internet telephony has advanced rapidly since then. Many software developers now offer PC telephony software but, more importantly, gateway servers are emerging an interface between the Internet and the PSTN (Phone switched telephone network). Equipped with voice-processing cards, these gateway servers enable users to communicate via standard telephones. The gateway server digitizes the analogue voice signal and compresses it into IP packets.

With its support for computer-to-telephone calls, telephone-to-computer calls and telephone-to-telephone calls, Internet telephony represents a significant step toward the integration of voice and data networks. It also offers tremendous cost savings, which is more than tempting for the standard freelance translator.

Internet telephony nevertheless still has some problems with reliability and sound quality, due primarily to limitations both in Internet bandwidth and current compression technology. This leads many companies to confine their Internet-telephony applications to their intranets. Whereas most translators working for big companies may have already been able to experience the advantages of this communication system, few freelancers have the same professional opportunity. Internet telephony within an intranet enables users to save on long-distance bills between sites; they can make point-to-point calls via gateway servers attached to the local-area network (LAN) and no PC-based telephony software or Internet account is required.

Let us image a case where Internet telephony is used in company intranets. User A in New York wants to make a point-to-point phone call to user B in the company's Geneva office. He picks up the phone and dials an extension to connect with the gateway server, which configures the private branch exchange (PBX) to digitize the upcoming call. User A then dials the number of the London office and the gateway server transmits the (digitized and IP-packetized) call over the wide-area network (WAN) to the gateway at the Geneva end. The Geneva gateway converts the digital signal back to analogue format and delivers it to the called party.

Technical barriers and limitations

Of course, there are important technical barriers to Voice-over-Internet Protocols. One of the main aims of Internet telephony as such is to achieve a reliable, high-quality voice service, which is the kind that users expect from phone switched telephone network. This is obviously a very important issue when talking about specific communication or about the transmission of important data, as in the case of a conversation where translation problems or terminology issues are dealt with. If the lack of reliability requires constant repetition or rephrasing in order to complete the process, this will lead translator to choose traditional slower or more expensive methods to solve doubts.

At the moment, the level of reliability of sound quality on the Internet has not reached its peak. This is mainly due to bandwidth limitations, which lead to packet loss. Packet loss usually triggers the undesirable appearance of gaps or periods of silence in the conversation, which produce a clipped-speech effect, clearly unsatisfactory not only for most standard users but for any kind of business interaction. For interaction among freelance translators and their informants. this kind of interference would be totally unacceptable.

The problem is probably due not so much to a lack of technological developments, but to the increasing popularity of Internet, with millions of new users signing on every month. This heavy, almost uncontrolled use of the limited bandwidth available usually results in congestion, which can also cause delays in packet transmission.

However, reliability and sound quality do not depend on bandwidth alone, since they are also determined by the voice-encoding techniques and associated voice-processing functions of the gateway servers. There are a great variety of speech-compression protocols. They all have their own speech-coding algorithms, as well as different bit rates and mechanisms for reconstructing voice packets. Depending on the quality of the algorithms used, there will be varying levels of intelligibility and fidelity in sound.

Present strategies to overcome difficulties

The industry is addressing all these problems with two main strategies: one of them consists of working on bandwidth limitations, which will be done by upgrading the Internet backbone to asynchronous transfer mode (ATM), a special system designed to handle voice, data and video packet loss. On the other hand, there have been several standard-setting efforts whose main aim is to focus on the three central elements of Internet telephony: the audio codec format, transport protocols and directory services.

The adoption of an audio codec standard has been a complicated process. The industry has agreed to sacrifice some sound quality for the sake of greater bandwidth efficiency. However, the main problem is that this will probably improve reliability and sound quality mostly for intranet traffic or point-to-point IP connections.

The current transport protocol still does not have mechanisms for ensuring the on-time delivery of traffic signals or for recovering lost packets. Neither does it address the quality of service issue, which aims at guaranteeing bandwidth availability for specific applications. In the near future, a new protocol may be adopted to improve quality-of-service levels.

Industry standards for directory services are also extremely important as they ensure the interoperability between the Internet and the PSTN.

The future of VoIPs

The future of Voice-over-Internet protocols is still to be seen but we can expect developments beyond the areas where they have been working up to now: corporate intranets and commercial extranets.

This development depends on the VoIP gateways, which evolve from PC-based platforms to robust embedded systems, each of which will virtually be able to handle hundreds of simultaneous calls, consequently reducing the expenses associated to high-volume voice, fax and videoconferencing traffic. IP will act as a unifying agent, concentrating all traffic, data, voice and video.

However, if we want the public Internet to handle voice and video services in a reliable manner, we definitely need two critical changes to take place: The improvement and deployment of the IP/ATM/synchronous optical network and ISDN, cable modems and x digital subscriber line (xDSL) technologies. We might also expect a certain segmentation of the public Internet, in which users will have to pay for the specific service levels they require.

Whereas the first critical change seems quite feasible, the second aspect is more problematic, since Internet users will normally not be willing to pay for a service that was previously free. Nevertheless, a cost-benefit analysis shows that a little investment will yield a high-quality service, and this will help us improve the quality of our work.

Videoconferencing and data collaboration are bound to become the normal method of corporate communications. Companies are aware of the economics of telecommuting, especially since network performance and interoperability are gradually increasing. Sooner than we expect, the video camera will become a standard piece of computer hardware for full-feature multimedia systems.

Applications to translation and interpreting

How can all these improvements improve our working conditions as translators? One of our main problems is the constant feeling of isolation. This may have its positive effects, but it certainly prevents us from sharing our successes and failures in an active and efficient manner. Breaking down communication barriers by means of VoIPs may help us to defeat our much maligned invisibility. Real-time interaction may also change the current state of the interpreter's work and crucial role.

That said, we still lack objective criteria to assess not only the efficiency but also the applicability of these tools to our work. This means that the success of a given communication system does not depend that much on the quality of the technology, but on the translator's ability to learn how to use and how to integrate real-time interaction with other colleagues or informants. Most of us are still used to paper documentation, and only a limited number of professionals either use or help build Internet glossaries or information sources. This must cast some doubt on our willingness to accept new technologies.

Another important problem, especially in the case of real-time video interaction, is that it is still absolutely compulsory that all the parties involved have the necessary software and functional Internet connection. In other words, there is no use in us translators trying to follow the breathtaking pace of technological development by buying and testing any new tool if our customers or information providers do not do the same. This again dangerously strengthens traditional or standard communication methods.

Nevertheless, the development of new gateway servers should allow us to phone our costumer or informant, who may only have a standard telephone line, from our computer with considerable savings and without moving from our computer desk. In the case of written translation, audiovisual communication is not an essential requirement: sound will be enough in most cases. Unfortunately this does not apply to interpreting, where the visualization of the presentation or the body language of the speaker is thought to be absolutely essential to guarantee high-quality service.

Final remarks

Beyond technical requirements, which are not really that problematic, there is certainly a need to test in a objective manner how these tools will improve the translator's efficiency and if this can be bring about time and money savings. Then again, objectiveness is hard to obtain when not all translators are keen on using new technologies or may not be able to find their right application if not previously guided or advised. Efficiency is also hampered by the fact that VoIPs are far from being well known by many Internet users.

One possibility could be to start working on big companies or institutions where financial and technical issues may be more limited. Alternatively, one could focus on the field of remote interpreting, trying to establish a series of parameters whereby we could both objectively and subjectively assess the effect of real-time Internet communication on the interpreter's performance. We could try to guarantee objectiveness by comparing the quality of the interpreter's speech in a standard working situation (booth located at the conference site) to that in a situation where the interpreter sees and receives the sound via Internet from a distant location. The interpreter could also fill in a questionnaire with their own subjective considerations with regard to issues such as sound and image quality, differences in the degree of comfort and security, and feelings stemming from the lack of real interaction or closeness to the lecturer or audience.

This is just an example of the research potential that Internet communications have in the field of translation and interpreting.