Voice Over IP Solutions

Enhance your broadband Internet service by offering your customers extra cost savings on their voice calls using advanced Internet telephony solutions.

Summary

The idea of using the Internet to make cheap telephone calls has been a technical reality for many years. But it has not achieved the same kind of popularity that other Internet applications like instant messaging, web cams and email have attained. The main reason for this has been the lack of high speed Internet connections that can deliver a voice call with the same quality of service and reliability we are used to with the traditional telephone networks.

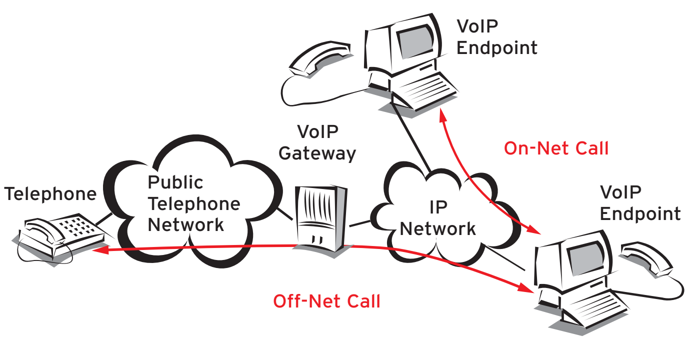
But that is now changing. The availability of high speed broadband Internet connections means that small business and home workers now have the high speed pipes that can guarantee voice quality. Also, in a highly competitive telecommunications environment, providers are looking for new features to differentiate their services from the crowd. Add these together and the time is now right for voice over IP to become a reality.

In this SMC business solutions brochure, we will explain how end users can benefit from a voice over IP solution to save money on telephone calls. For Internet service providers and broadband sellers, voice over IP can provide a new attractive service that will attract new customers in a highly competitive market. Not all telephone calls are appropriate for a voice over IP solution; we will describe sweet spots of the market where, the solution is appropriate. Briefly, we will explain how voice over IP works and the key technologies involved. Finally, we will demonstrate how working with SMC, you can implement an Internet telephony solution to save money on your calls or give your broadband customers an extra reason for choosing your service.

Voice over IP implementation

Voice over IP, or VoIP as it is commonly called, can be deployed and implemented in many different ways and for different purposes. There are two main categories of VoIP calls: on-net and off-net as shown in figure 1. An on-net call originates and terminates with users attached to a single enterprise or provider network. Similarly off-net calls on the other hand cross the boundaries of IP networks and traditional public telephone networks and require special gateways to connect and convert the encoded calls.

Figure 1: Voice over IP scenarios



Voice over IP solutions

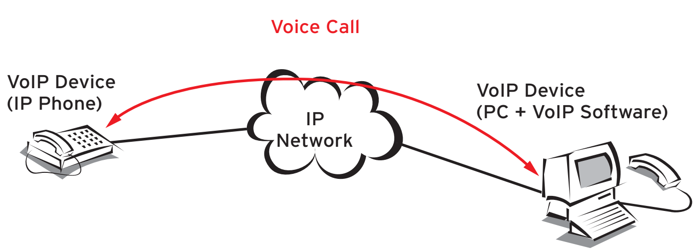
Voice over IP is a technology that can be used to provide solutions to solve different problems. It should not be considered as a replacement for the traditional telephone system which works extremely well in most situations. Rather, voice over IP can be used to implement the following services:

- IP Telephony
- Internet Telephony
- LAN Telephony

IP telephony

The basic idea of IP telephony is to connect to voice over IP devices using an IP network and make a telephone call between them as shown in figure 2. The end devices may be special VoIP telephone handsets or a computer running suitable VoIP software.

Figure 2: IP telephony



Internet telephony

An Internet Telephony Service Provider (ITSP) interconnects the public telephone network with the Internet by implementing interconnection gateways. Residential users as well as enterprise users can make very cheap (sometimes cost free) on-net calls within the provider IP network, or over the Internet.

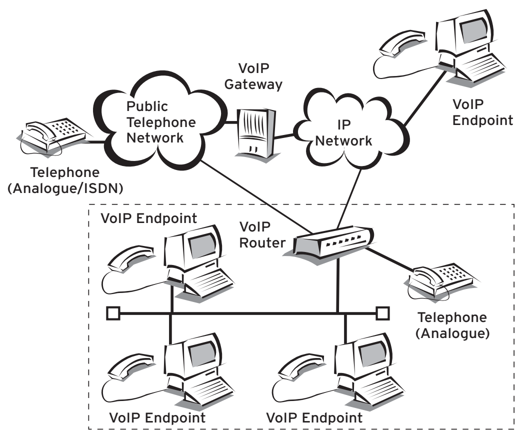
Most ITSPs offer telephony services that allow their users/subscribers to establish telephone communications also with subscribers that are connected to the PSTN (off-net calls) at comparatively low charges.

Subscribers are usually assigned a normal telephone number by the ITSP. Every call that is made from the PSTN to that number will be routed by the interconnection gateway to the customer's IP phone.

The same technology approach is the basis for special long distance call service offerings where an IP network infrastructure is used to interconnect two PSTN subscribers. Significant cost reduction for long distance calls are often possible in these cases. This is sometimes referred to as toll-bypass.

A typical application is to allow small business or home workers to use their broadband internet connection for making voice calls as shown in figure 3. End devices can be: VoIP telephones, VoIP software equipped PCs, or ordinary analogue telephones attached directly to the user's VoIP router.

Figure 3: Internet telephony

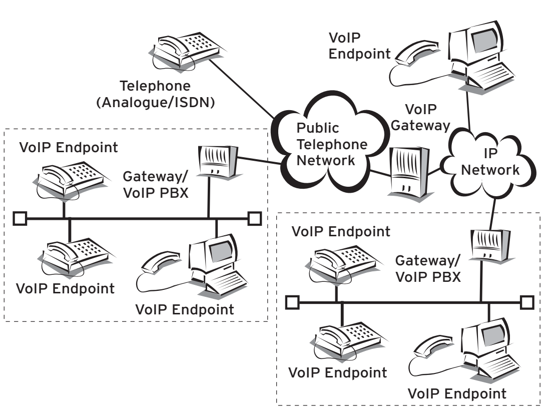


For instance, the SMC7908VoWBRA/B Voice Connect Wireless ADSL gateway can be used as the VoIP router in Figure 3. It is easy to set up and can support 1 or 2 analogue telephones plugged directly into the routers' voice ports, or VoIP software clients installed on the local PCs. No need to invest in expensive IP phones. Calls on the service provider's network are typically free and offnet calls are normally charged at a lower rate than most fixed line calls.

LAN telephony

A LAN telephone solution is more appropriate for medium to large enterprises that wish to use their existing IP network to save money on voice calls. Calls can be made locally using the existing LAN, or remotely across their private IP network as shown in figure 4.

Figure 4: LAN telephony



A business scenario for Voice over IP

Voice over IP is not guaranteed to save money in every situation. It is important to look at the local telephone tariffs in order to position the service appropriately. For instance, in many countries national calls are charged at a fixed rate and competitive with local calls. VoIP is unlikely to offer significant savings in these areas. However, markets where there are many inter-carrier or international calls are likely to have higher call charges. An IP or Internet telephony solution could offer significant savings in these markets.

Most Internet telephony service providers will offer on-net calls for free between VoIP users connected to the same network. Off-net calls are offered at reduced rates as the ITSP is able to buy capacity at wholesale rates and is usually able to negotiate a better deal than the normal end-user tariffs.

For example, table 1 shows the typical call charges from Belgium, Netherlands, Germany and UK using traditional telephone carriers and a typical VoIP provider.

Table 1: Telephone call charges (Ä cents for a 10 minute call)*

Called Country:	Belgium		Netherlands		Germany		UK		USA	
	PSTN	VoIP	PSTN	VoIP	PSTN	VoIP	PSTN	VoIP	PSTN	VoIP
Belgium	52	37	187,3	60	187,3	60	187,3	60	187,3	60
VoIP Savings	29%		68%		68%		68%		68%	
Netherlands	81,4	50	43,9	28	74,5	29	71,6	50	74,5	50
VoIP Savings	39%		36%		61%		30%		33%	
Germany	123	30	123	27	120	17,9	123	26	123	23
VoIP Savings	76%		78%		85%		79%		81%	
UK	71	29	71	26	71	24	43	17	71	21
VoIP Savings	60%		64%		66%		60%		70%	

As can be seen, depending on the country where the call originates from, savings of over 80% on both national and international calls are possible.

* These calculations are examples found in the countries noted above in January 2005 and should be used as a guide only.

How does Voice over IP work?

There are three basic protocols which are used to implement a voice over IP solution: H.323, SIP and RTP. H.323 and SIP are mainly concerned with the call establishment and voice encoding, whereas RTP (Real-time Transport Protocol) is used by both protocols for the transport of encoded voice packets over an IP network.

H.323

H.323 is an interoperability standard that describes the modes of operation required for various audio, video, and/or data terminals to work together. It is one of the most important standards for IP voice and video applications including IP phones as well as audio and video conferencing equipment.

The development of H.323 was originally focused on LAN video conferencing applications. When it was ratified in 1996 by the ITU (International Telecommunication Union), it became an umbrella standard, defining a broad range of audio, video and data compression standards and protocols.

SIP

The Session Initiation Protocol (SIP) is a relatively new Internet standard that specifies a simple signaling and application layer control protocol for multimedia conferencing and telephony. It is today along side H.323, the most important standard for IP and Internet voice applications.

SIP is defined in RFC 2543, as part of the MMUSIC (Multiparty Multimedia Session Control) working group of the IETF. SIP is a simple, low-level protocol for establishing, modifying or terminating multimedia sessions or Internet telephony calls between two or more users. Such sessions can include voice, video, chat, other multimedia data (such as interactive games). The protocol can also invite participants to unicast or multicast sessions.

SIP supports name mapping and redirection services. This makes it possible to identify users regardless of where they are and allows users to initiate and receive connections and services from any location. This feature is important for supporting traveling workers using mobile wireless networks.

SIP is a simple request-response protocol, dealing with requests from clients and responses from servers. Users are identified by SIP URLs. SIP determines the end system to be used for the session, the communication media and media parameters. It then initiates the communication. It establishes call parameters at each end of the communication, and handles call transfer and termination. Like H.323 SIP uses the RTP (Real-Time Transport Protocol) for transporting the voice (multimedia) information.

SMC's Voice over IP products

SMC Networks has over 30 years experience in designing and manufacturing network equipment. SMC's Internet solutions include: Ethernet switches, ADSL routers, voice gateways, network interface cards plus a complete range of wireless routers, access points, adapter cards, bridges and antenna.

The SMC7908VoWBRA/B is designed for use on ADSL Annex A (SMC7908VoWBRA) networks.

VoIP Features

- SMC7908VoWBRA:
  - 1x FXO (RJ-11) for automatic link fail-over to PSTN,
  - 1x FXS (RJ-11) for connecting an analogue telephone
- SMC7908VoWBRB: 2x FXS (RJ-11) for connecting analogue telephones
- Support for both the SIP and H.323 protocols (different firmwares)
- PSTN supplementary services:
  - Call Forward
  - Call Hold
  - Call Waiting
  - Call Transfer
  - Caller ID
- Support for multiple voice CODECs:
  - G.711 A/U Law
  - G.729a
  - G.723.1
- T.38 Fax relay and modem relay
- QoS process with ToS and Diffserv
- Echo cancellation
- Jitter buffer
- Mapping of voice and data to separate PVC's
- DTMF tone generation and relay
- Dynamic pass-through for SIP/H.323 client on LAN

Barricade™ Voice Connect Wireless ADSL Gateway

The SMC7908VoWBRA is a complete all-in-one solution for your small or home office. It combines a broadband wireless router with a Voice over IP gateway giving you an integrated voice and data office network in one box. You can share your broadband Internet connection with local PCs through the built-in 4 port Ethernet hub or wirelessly using the 54Mbps wireless access point. Plus you can plug in a telephone and save money by calling friends and colleagues over the Internet using the unique Voice over IP gateway.

For more information

For more information about SMC's unique voice, broadband and wireless solutions and to find out how they can offer broadband users cost savings on their voice telephone calls, contact your local SMC distributor/reseller or visit our website at www.smc.com to locate your nearest SMC distributor/reseller.