VoIP: Innovation In Telecommunications!

VOIP stands for Voice Over Internet Protocol. It is an innovative way to talk with others around the world. VOIP has done amazing things for phone conversations and telephone conferences that take place with companies in other countries. The connection is so clear you won’t believe how far away the other party is.

The VOIP process uses the internet connection to route phone calls. In addition to offering the highest clarify in your calls, you will find the cost to be much less than normal long distance services. For those who make a lot of calls outside of their area code, this is a great savings. In fact, the service allows many long distance calls to be billed as local calls instead.

The VOIP is a great option for those on the go for business or leisure. You can get incoming calls anywhere you can get an internet connection. As more internet ports are being added all the time to various locations, the VOIP will soon be usable everywhere you can image.

One disadvantage of the VOIP system is that is won’t operate during a power outage, and that may be a time when you need it for emergency calls. Some VOIP systems are offering a backup battery for this type of situation. Because the underlying IP network is inherently unreliable, in contrast to the circuit-switched public telephone network, and does not inherently provide a mechanism to ensure that data packets are delivered in sequential order, or provide Quality of Service (QoS) guarantees, VoIP implementations face problems mitigating latency and jitter. This is especially true when satellite circuits are involved, due to long round-trip propagation delay (400–600 milliseconds for links through geostationary satellites). The receiving node must restructure IP packets that may be out of order, delayed or missing, while ensuring that the audio stream maintains a proper time consistency. This function is usually accomplished by means of a jitter buffer in the voice engine. Another challenge is routing VoIP traffic through firewalls and address translators. Private Session Border Controllers are used along with firewalls to enable VoIP calls to and from protected networks. Skype uses a proprietary protocol to route calls through other Skype peers on the network, allowing it to traverse symmetric NATs and firewalls. Other methods to traverse firewalls involve using protocols such as STUN or ICE. Some broadband connections may have less than desirable quality. Where IP packets are lost or delayed at any point in the network between VoIP users, there will be a momentary drop-out of voice. This is more noticeable in highly congested networks and/or where there are long distances between end points. Technology has improved the reliability and voice quality over time and will continue to improve VoIP performance. It has been suggested to rely on the packetized nature of media in VoIP communications and transmit the stream of packets from the source phone to the destination phone simultaneously across different routes (multi-path routing). In such a way, temporary failures have less impact on the communication quality. In capillary routing it has been suggested to use at the packet level Fountain codes or particularly raptor codes for transmitting extra redundant packets making the communication more reliable.

The VOIP phones offer various payment plans to fit your phone use needs. The VOIP pre-paid phone card is very popular. While this is not an actual phone, it allows the user to make the same affordable calls, from anywhere in the world - right from their computer. The pre-paid phone card is a great product for those working or traveling in another country.

It is believed the VOIP will soon replace many home phones and cell phones. An individual will use their VOIP for both. This is because the cost will be less and the service will be better.

About the Author

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