SO WHAT IS VOIP........?

VoIP is short for voice-over-Internet Protocol. The first question that comes to mind is "What is Internet Protocol (IP)?" And the next question is "Why should you care about VoIP?" To answer the first, a traditional private branch exchange (simply called PABX) involves only ISDN or analogue lines.

The Internet Protocol is a means by which computers communicate with each other. IP PBX (Internet Protocol Private Branch eXchange) is a telephone switching system that allows people to speak internally within a local network (LAN), their wide area network (WAN) as well as externally to other standard phone lines. Those who use an IP PBX will be able to make VoIP-to-VoIP, VoIP-to-ISDN, VoIP to analogue, ISDN to ISDN or ISDN to-analogue calls…. so the IP PBX is a newer and more flexible technology. So why shouldn't voice or telephones communicate using the same protocol? The answer to the second question is simple: applications.

HOW IT WORKS

IP PBX works by converting IP phone calls into traditional circuit-switched calls, thereby combining voice and data communications into one line. A traditional PBX, however, must send voice and data communications in two separate lines. With the lines converged, those using an IP PBX will be able to access the Internet, speak on a VoIP telephone and speak on a variety of handsets using the same network line. The transmission of your voice looks no different to the network than other information (data), being passed along on your corporate infrastructure. The term used is to packetise your voice, which conjures up the idea of a nice neat little package, and indeed it is.

In every IP telephone there is a device called a NIC or network interface card (a small printed circuit board). Your packetised voice is sent through the NIC onto the network using the same pair of wires and through the same routers and LAN switches as your data communications.

WHERE THIS LEADS US........

VoIP has joined the World Wide Web (www) and e-mail as the third ubiquitous and critical Internet application and the next decade is set to prove this. With South Africa's current proliferation and market readiness in terms of VoIP adoption and development, Xtranet and Xtravoice will provide solutions that are cost effective, flexible and enable the future of this technological advancement.

Xtranet and Xtravoice focuses on the delivery and implementation of converged telecommunications solutions via open source platforms and we are able to deliver and implement converged telecommunications solutions ranging from PBX and switching to a full Voice over Internet Protocol (VoIP) platform that is reliable, flexible and cost effective

Xtranet and Xtravoice provide businesses with an evolutionary means of exploiting the benefits of IP-Telephony and VoIP through our PBX and VoIP gateways. LCR is by definition, the routing of voice calls along the most cost effective route; whether that call originates from a Voice Over Internet Protocol (VoIP), "Telkom" like carrier or "Cellular" backbone that manages and makes routing decisions based on call destinations and capacity availability to effectively drive down telephony costs

VoIP is expected to become the standard LCR routing of all premium calls via Internet Protocol based networks to centrally hosted Value Added Network Service Provider (VAN) services and, ultimately, providing a complete telecommunications alternative to businesses and consumers to traditional "Telkom" delivered services.

This fundamental shift in the very definition of LCR will likely have a major impact on the local telecommunications sector – leading to consolidation within the LRC market and expansion in the VoIP arena. The LCR Solution allows your company to implement cost savings on national, International, Mobile and inter-branch voice calls maximizing your investment and productivity.

IP-PBX's and VoIP gateways are programmed to recognise numbers called - will automatically divert these calls through a fixed Internet connection to any voice network/ terminal worldwide thus effectively converting traditional "Telkom" based calls to inexpensive hybrid Internet calls which can show your organisation substantial cost savings.

VoIP:

VoIP (Voice over Internet Protocol) is a general term for a family of transmission technologies for delivery of voice communications over IP networks such as the Internet or other packet-switched networks.

Codec is a device or computer program capable of encoding and/or decoding a digital data stream or signal. The word *codec* is a blending of two words '**co**der-**dec**oder'. Historically a modem was a contraction of modulator/demodulator (modem was called dataset by telcos) and converted digital data from computers to analog for phone line transmission. On the receiving end the analog was converted back to digital. CODECs did the opposite (convert audio analog to digital (01110011) and then computer digital sound back to audio). There was no compression involved in CODECs, only coding and decoding.MP3 and WAVE is eg of codec's.

SUPPORTED CODEC'S:

G.711, also known as Pulse Code Modulation (PCM), is a very commonly used waveform codec. G.711 uses a sampling rate of 8,000 samples per second, with the tolerance on that rate 50 parts per million (ppm). Non-uniform quantization (logarithmic) with 8 bits is used to represent each sample, resulting in a 64 kbit/s bit rate. There are two slightly different versions; μ-law, which is used primarily in North America, and A-law, which is in use in most other countries outside North America.

G.722[1] is an ITU-T standard 7 kHz wideband speech codec operating at 48, 56 and 64 kbit/s. Technology of the codec is based on sub-band ADPCM (SB-ADPCM).G.722 sample audio data at a rate of 16 kHz (using 14 bits), double that of traditional telephony interfaces, which results in superior audio quality and clarity.

G.723 is an ITU-T standard speech codec. This is an extension of Recommendation G.721 adaptive differential pulse code modulation to 24 and 40 kbit/s for digital circuit multiplication equipment (DCME) applications.

G.726 is an ITU-T ADPCM speech codec standard covering the transmission of voice at rates of 16, 24, 32, and 40 kbit/s. The most commonly used mode is 32 kbit/s, which doubles the usable network capacity by using half the rate of G.711. It is primarily used on international trunks in the phone network. The principal application of 24 and 16 kbit/s channels is for overload channels carrying voice in digital circuit multiplication equipment (DCME). The principal application of 40 kbit/s channels is to carry data modem signals in DCME, especially for modems operating at greater than 4800 kbit/s.It also is the standard codec used in DECT wireless phone systems and is used on some Canon cameras.

G.729 is an audio data compression algorithm for voice that compresses digital voice in packets of 10 milliseconds duration. It is officially described as *Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)*. Because of its low bandwidth requirements, G.729 is mostly used in Voice over Internet Protocol (VoIP) applications (such as Skype) where bandwidth must be conserved. DTMF tones, Fax transmissions, and high-quality audio cannot be transported reliably with this codec. DTMF requires the use of the RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals as specified in RFC 2833

GSM (Global System for Mobile Communications): is the most popular standard for mobile telephony systems in the world

iLBC (Internet Low Bitrate Codec) is a royalty-free narrowband speech codec . It is suitable for VoIP applications, streaming audio, archival and messaging. The algorithm is a version of block-independent linear predictive coding, with the choice of data frame lengths of 20 and 30 milliseconds. The encoded blocks have to be encapsulated in a suitable protocol for transport, usually the Real-time Transport Protocol (RTP). iLBC handles the case of lost frames through graceful speech quality degradation. Lost frames often occur in connection with lost or delayed IP packets. Ordinary low-bitrate codecs exploit dependencies between speech frames, which unfortunately results in error propagation when packets are lost or delayed. In contrast, iLBC-encoded speech frames are independent and so this problem will not occur.

Linear is a royalty-free narrowband speech codec, developed by Global IP Solutions (GIPS) formerly Global IP Sound. It is suitable for VoIP applications, streaming audio, archival and messaging. The algorithm is a version of block-independent linear predictive coding, with the choice of data frame lengths of 20 and 30 milliseconds. The encoded blocks have to be encapsulated in a suitable protocol for transport, usually the Real-time Transport Protocol (RTP).

LPC10is a secure telephony speech encoding standard developed by the United States Department of Defense and later by NATO. The standard was finished 1984. The algorithm used is a linear predictive coding vocoder. The vocoder enables understandable speech, but the quality is very unnatural and synthetic. File size is 20 times smaller than MP3 - it is very small.

Speex is a patent-free audio compression format designed for speech and also a free software speech codec that may be used on VoIP applications and podcasts. It is based on the CELP speech coding algorithm. Speex claims to be free of any patent restrictions and is licensed under the revised (3-clause) BSD license. It may be used with the Ogg container format or directly transmitted over UDP/RTP.

PROTOCOLS:

Protocol is the set of standard rules for data representation, signaling, authentication and error detection required to send information over a communications channel. An example of a simple communications protocol adapted to voice communication is the case of a radio dispatcher talking to mobile stations. Communication protocols for digital computer network communication have features intended to ensure reliable interchange of data over an imperfect communication channel. Communication protocol is basically following certain rules so that the system works properly. A protocol makes the decision what codec

To use and through which port in the firewall eg Browsing will go through Port 80

H.323 is a recommendation from the ITU Telecommunication Standardization Sector (ITU-T) that defines the protocols to provide audio-visual communication sessions on any packet network. The H.323 standard addresses call signaling and control, multimedia transport and control, and bandwidth control for point-to-point and multi-point conferences. It is widely implemented by voice and videoconferencing equipment manufacturers, is used within various Internet real-time applications such as GnuGK and NetMeeting and is widely deployed worldwide by service providers and enterprises for both voice and video services over Internet Protocol (IP) networks. H.323 Call Signaling is based on the ITU-T Recommendation Q.931 protocol and is suited for transmitting calls across networks using a mixture of IP, PSTN, ISDN, and QSIG over ISDN. A call model, similar to the ISDN call model, eases the introduction of IP telephony into existing networks of ISDN-based PBX systems, including transitions to IP-based Private Branch exchanges (PBXs).

IAX (Inter-Asterisk eXchange) protocol is native to Asterisk PBX and supported by a number of other softswitches and PBXs. It is used for enabling VoIP connections between servers beside client-server communication. IAX now most commonly refers to IAX2, the second version of the IAX protocol. The original IAX protocol has been deprecated in favor of IAX2 IAX will not choose random ports like SIP and will only send on SCTP this makes this the preferred Protocol in SA due to our bandwidth constraints

SIP (Session Initiation Protocol) is a signaling protocol, widely used for controlling multimedia communication sessions such as voice and video calls over Internet Protocol (IP). The protocol can be used for creating, modifying and terminating two-party (unicast) or multiparty (multicast) sessions consisting of one or several media streams. The SIP protocol is a TCP/IP-based Application Layer protocol. SIP is designed to be independent of the underlying transport layer; it can run on Transmission Control Protocol (TCP), User Datagram Protocol (UDP), or Stream Control Transmission Protocol (SCTP).It is a text-based protocol, incorporating many elements of the Hypertext Transfer Protocol (HTTP) and the Simple Mail Transfer Protocol (SMTP), allowing for direct inspection by administrators.

MGCP (Media Gateway Control Protocol) for controlling Media Gateways on Internet Protocol (IP) networks and the public switched telephone network (PSTN). MGCP is a signaling and call control protocol used within Voice over IP (VoIP) systems that typically interoperate with the public switched telephone network (PSTN). As such it implements a PSTN-over-IP model with the power of the network residing in a call control center (softswitch, similar to the central office of the PSTN) and the endpoints being "low-intelligence" devices, mostly simply executing control commands.

SCCP (**Signalling Connection Control Part**) is a network layer protocol that provides extended routing, flow control, segmentation, connection-orientation, and error correction facilities in Signaling System 7 telecommunications networks. SCCP relies on the services of MTP for basic routing and error detection.

PPP (Point-to-Point Protocol) is a data link protocol commonly used to establish a direct connection between two networking nodes. It can provide connection authentication, transmission encryption privacy, and compression. PPP is used over many types of physical networks including serial cable, phone line, trunk line, cellular telephone, specialized radio links, and fiber optic links such as SONET. Most Internet service providers (ISPs) use PPP for customer dial-up access to the Internet.

PPPoE (Point-to-Point Protocol over Ethernet) is a network protocol for encapsulating Point-to-Point Protocol (PPP) frames inside Ethernet frames. It is used mainly with DSL services where individual users connect to the DSL modem over Ethernet and in plain Metro Ethernet networks.

Network and communications:

PoE (Power over Ethernet) describes a system to safely pass electrical power, along with data, on Ethernet cabling. PoE requries category 5 cable or higher for high power levels. Power can come from a power supply within a PoE-enabled networking device such as an Ethernet switch or from a device built for "injecting" power onto the Ethernet cabling, dubbed Midspan.

DSL (Digital Subscriber Line) is a family of technologies that provides digital data transmission over the wires of a local telephone network. DSL originally stood for *digital subscriber loop*. In telecommunications marketing, the term Digital Subscriber Line is widely understood to mean Asymmetric Digital Subscriber Line (ADSL), the most commonly installed technical varieties of DSL. DSL service is delivered simultaneously with regular telephone on the same telephone line as it uses a higher frequency band that is separated by filtering. The data throughput of consumer DSL services typically ranges from 384 KB/s to 20 MB/s in the direction to the customer, depending on DSL technology, line conditions, and service-level implementation. Typically, the data throughput in the reverse direction, i.e. in the direction to the service provider is lower, hence the designation of *asymmetric* service, but the two are equal for the Symmetric Digital Subscriber Line (SDSL) service.

Last mile is the final leg of delivering connectivity from a communications provider to a customer. The phrase is therefore often used by the telecommunications and cable television industries. The actual distance of this leg may be considerably more than a mile, especially in rural areas. It is typically seen as an expensive challenge because "fanning out" wires and cables is a considerable physical undertaking. Because the last mile of a network to the user is also the first mile from the user to the world, the term "*first mile*" is sometimes used. To solve the problem of providing enhanced services over the last mile, some firms have been mixing networks for decades. One example is Fixed Wireless Access, where a wireless network is used instead of wires to connect a stationary terminal to the wireline network. Various solutions are being developed which are seen as an alternative to the "last mile" of standard incumbent telecommunications providers: these include WiMAX and BPL (Broadband over Power Line) applications.

Leased line is a symmetric telecommunications line connecting two locations. It is sometimes known as a 'Private Circuit' or 'Data Line' in the UK or as CDN (Circuito Diretto Numerico) in Italy. Unlike traditional PSTN lines it does not have a telephone number, each side of the line being permanently connected to the other. Leased lines can be used for telephone, data or Internet services. Some are ringdown services, and some connect two PBXs. A permanent telephone connection between two points set up by a telecommunications common carrier. Typically, leased lines are used by businesses to connect geographically distant offices. Unlike dial-up connections, a leased line is always active. The fee for the connection is a fixed monthly rate. The primary factors affecting the monthly fee are distance between end points and the speed of the circuit. Because the connection doesn't carry anybody else's communications, the carrier can assure a given level of quality.

Call Termination, also known as voice termination, refers to the handing off or routing of telephone calls from one telephone company, also known as a carrier or provider, to another. This term often applies to calls while using Voice over Internet Protocol (VoIP): a call initiated as a VoIP call is terminated using the public switched telephone network (PSTN). In such cases, termination services may be sold as a separate commodity. The opposite of call termination is call origination, in which a call initiated from the PSTN is terminated using VoIP. Thus, in "origination" a call originates from PSTN and goes to VoIP, while in "Termination" a call originates in VoIP and terminates to PSTN.

FXS (Foreign exchange Station) is a telephone interface which supplies battery power, provides dialtone, and generates ringing voltage. A device that connects to such an interface contains a Foreign exchange office (FXO) interface and could be a standard analog telephone or a private branch exchange (PBX) to receive telephone service. Any telephone exchange is an example of an FXS, as is the telephone jack on the wall, though the term is rarely applied except in connection with foreign exchange service. An FXS interface utilizes an FXO protocol to detect when the terminating device (telephone) goes on-hook or off-hook, and can send and receive voice signals. An FXS interface provides service at the "station" end of a foreign exchange line.

FXO (foreign exchange office) designates a telephone signaling interface that receives POTS, or "plain old telephone service". It generates the off-hook and on-hook indications (loop closure/non-closure) at the FxS's end of a telephone circuit. Analog telephone handsets, fax machines and (analogue) modems are FXO devices, though the term is rarely used except in connection with Foreign exchange service (FX). FXO interfaces are also available for computers and networking equipment, to allow these to interact directly with POTS systems. These are commonly found in devices acting as gateways between Voice over Internet Protocol (VoIP) systems and the public switched telephone network (PSTN).

ATA (analog telephony adapter, or analog telephone adapter) is a device used to connect one or more standard analog telephones to a digital and/or non-standard telephone system such as a Voice over IP based network.