Telecommunications Glossary

API - Application Programming Interface. An API allows two applications to communicate. It's what enables data to be seamlessly distributed to different applications on different devices in different locations, and then updated and manipulated in real time. They're at the heart of cloud-based Unified Communications.

Asynchronous Transfer Mode - A high-speed networking standard for voice and data. It's typically used for private long-distance networks. ATM utilizes fixed-sized cells rather than the variable-length packets Ethernet utilizes. This makes it easier to manage bandwidth; however, it is typically more expensive.

Auto Attendant - Basically, this is a virtual receptionist. Auto attendant allows callers to be automatically transferred to an extension without the intervention of an operator. They typically have a simple menu system and are customizable.

Availability - Availability is based on the probability of a hardware failure, which is obviously bad. Availability is determined by dividing the "mean time between failure" by the "mean time to repair" - in other words, how often things break down divided by how long it takes to fix them. "Five nines," i.e. 99.999 percent, is the benchmark people often talk about.

Bring-your-own-device (BYOD) - this is a huge reality for most businesses today - their workforce is committed to using its personal smartphones, tablets and other mobile devices to access business information and applications. On one hand, this is great, because it means employees can work anywhere and offers the promise of increased productivity. On the other hand, data governance becomes a major issue. A successful BYOD initiative requires a robust technology platform, good security and proper policy in place.

CALEA - *The Communications Assistant for Law Enforcement Act*. This relates to electronic surveillance, which has been a very hot topic since 2013. The act requires that carriers and manufacturers enable their equipment, facilities, and services to ensure that they have built-in surveillance capabilities so that federal agencies can monitor communications.

CPNI - This stands for *Customer Proprietary Network Information*. This is the data collected about an individual user's calls, such as time, date, duration and destination number of each call.

Caller ID - Caller ID is used to allow the called party to see the calling party's name and telephone number before picking up the phone.

Click to Call - The ability to initiate a phone call from the contact list on your computer with the click of a mouse.

Cloud Communications - Voice and data communications over the Internet. All applications, switching and storage are hosted by a third party outside the organization and accessed over the Internet. With Cloud communications, there is no major capital expenditure for an in-house PBX system and ongoing costs are more predictable than with a traditional premise-based solution.

Computer Telephony Integrations (CTI) - Technology that enables integrated interaction on a telephone and a computer, such as click-to-call and screen pops.

Customer Relationship Management (CRM) - software that manages all aspects of an organization's interactions with customers and prospects. A unified communications solution can make a CRM system more accessible across an organization because it creates a heightened level of accessibility.

Denial of Service (DoS) - DoS attacks are favorites of hackers, criminals and other trouble-causing types. A DoS attack is an attempt to make a network unavailable to its intended users.

Direct Inward Dial (DID) - DID is used for call routing. Through DID, external callers are able to contact a user directly at his/her unique phone number.

Endpoint - An endpoint is an IP telephone, a softphone, or an analog telephone adapter device.

Extension - A standard extension is an individual user account on the cloud associated with a physical endpoint y a two to six-digit number. A

Cloud extension is not associated with a physical endpoint, i.e., a voice mailbox, etc.

Find me, Follow Me - Generally, this is used as a call-forwarding feature. It improves worker productivity and customer service by ensuring that every call reaches the right person, regardless of where he or she is working.

Firewall - A key security feature that you've almost certainly heard of. The firewall sits between two networks, such as a company's internal network and the Internet, and prevents unauthorized people from accessing the internal network.

Fixed Mobile Convergence (FMC) - This is exactly what it sounds like - the convergence of the fixed and mobile networks. FMC solutions integrate cellular services with private communications networks, whether they are wireless or wired.

Frame Relay - A cost-efficient method of data transmission for intermittent traffic between LANs and between end-points in WANs. They're less expensive than private leased lines because the carrier shares the frame relay bandwidth among many customers. This can have a negative impact on quality, and therefore requires detailed attention to engineering the solution.

Hosted VOIP - Also called Hosted. When we say "hosted" we mean that the hardware and PBX are hosted at an off-site location from where the VOIP telephone service is being used. Many businesses are embracing Hosted VOIP because it provides them robust communications, cost certainty and future-proofing of their business, while eliminating major capital expenditures for new in-house phone equipment.

Instant Messaging (IM) - Real-time communication over the Internet using text-based messages. Popular consumer-facing examples include G-chat, AIM and iMessage. IM is usually a central feature of unified communications.

IP PBX - This stands for Internet Protocol-Private Branch Exchange. It's a business phone system that delivers voice or video over a data network using IP.

IP Phones - IP phones plug directly into the network and perform analog-to-digital and/or digital-to-analog conversions.

IP Telephony - More commonly referred to as Voice over IP (VOIP). IP telephony uses the IP network to carry voice communications, replacing the public switched telephone network.

ITSP - Internet Telephony Service Provider. A company that provides VOIP services. We usually break them down into Interconnects, MSPs and Carriers.

Jitter - Generally cause by network congestion, which can create timing issues for when packets arrive, thereby contaminating voice calls and creating poor and/or unacceptable voice quality. This often manifests itself with "squawking" noises and other strange noises interrupting audio calls.

Latency - Latency is the time it takes for a caller's voice to be transported - packetized, sent over the network, de-packetized and replayed - to the other person. Too much latency is bad, making for a disjointed conversation flow. Ideally, latency should not exceed 100 milliseconds. Geographical distance or a lower-speed network connection can cause latency issues.

Managed IP Telephony Services - Hosted services. Typically, the end-customer business owns the IP PBX and related equipment

Mobile Device Management (MDM) - MDM software secures, monitors and manages mobile devices. It's used to enforce data and configurations for mobile devices, and includes the ability to lock or delete data from a lost or stolen mobile device.

Mobile Unified Communications (mobile UC) - A mobile UC solution seamlessly pulls together common telephony functions, voice, presence, chat, data, applications and other technologies from a smartphone or tablet.

Mobility Router - Allows users to make and receive calls from enterprise and personal mobile phone numbers by automatically selecting the best network (Wi-Fi or cellular) to optimize cost, call quality and batter life.

Origination & Termination - Origination refers to inbound calls or minutes from the PSTN. Termination refers to outbound calls or minutes from the PSTN.

Packets - In VOIP, voice is converted into data packets that are transmitted over the IP network and then reassembled into voice by an endpoint VOIP device.

Packet Loss - Packet loss occurs when one or more packets of data fail to reach their destination, resulting in a metallic sound or conversation dropouts. It can be cause by network congestion, distance and poor line quality. Excessive packet loss is perceived as broken or missing communication.

Post Dial Delay (PDD) - The interval between dialing the last digit of the called number and hearing the ring back tone.

Presence Status - Or just "presence" for short. This is the ability to see a colleague's presence status, whether he is in the office, away from his desk, etc.

Private Branch Exchange (PBX) - A term dating back to the days of switchboard operations, referring to an organization's telephone exchange.

Public Switched Telephone Network (PSTN) - The network of the public circuit-switched telephone lines, which allows any phone in the world to connect with any other phone.

Quality of Service (QoS) - A measure of the overall performance of the network. It takes into consideration factors such as error rates, bandwidth, throughput, transmission delay, availability and jitter.

Real-Time Transport Protocol (RTP) - An IP standard for delivering audio and video over IP networks.

Reliability - Reliability is determined by calculating how often the system fails compared to the percentage of the time that the system is available. The system should be available at least 99.999 percent of the time.

RespOrg - Stands for "Responsible Organization," the process of transferring "toll Free" telephone numbers from one carrier to another.

Service Level Agreement (SLA) - Part of a service contract that defines the level of service. An SLA for voice quality typically includes call completion rate, PDD, and some measure of voice quality.

Session Border Controller (SBC) - Network device used to register, set-up, control and tear down VOIP multimedia communications sessions. An SBC ensures that only approved traffic passed into the heart of your business. An SBC also hides your internal network and your users IP addresses from the outside world, which provides additional security protection.

Session Initiation Protocol (SIP) - SIP is a text-based signaling protocol used to control multimedia communication sessions over IP networks. SIP handles call controls such as ring, hold, transfer, conference, etc.

SIP Trunking - VOIP service used to deliver communication services to a customer's SIP-based PBX or endpoint, usually to a publically routable IP domain.

Softphone - A softphone is voice software that emulates a VOIP telephone on a computer, smartphone or tablet.

Trunk Line - Trunk lines connect the PBX to a public switched telephone network. With a Cloud-based solution, SIP trunks are used to make this connection.

Unified Communications (UC) - The seamless integration of voice, presence, chat, data, applications, and other technologies to improve communications, processes, and business productivity.

Unified Communications and Collaboration (UCC) - UCC integrates multiple communications channels to enable communications and collaboration across a company.

Unified Messaging - The integration of email, SMS, fax, voicemail and video messaging into a single interface which is accessible from a variety of devices. It simplifies user experience because all types of messages are stored in one place.

Virtual LANs (VLANs) - Used to "logically separate" devices and departments on the same Ethernet wire.

Virtual Private Network (VPN) - A private network extended across a public network, i.e. the Internet. It enables the sharing of data across public networks in a typically more secure and functional way.

Virtualization - Allows companies to consolidate servers, increase operational flexibility and deliver higher application availability. Virtualization utilizes a group of servers in a way that allows for running multiple operating systems simultaneously on the same machine.

VOIP (VoIP) - Voice calls over the IP data network. VOIP converts analog voice signals into digital data packets and supports real-time, two-way transmission of conversations using Internet Protocol.

WebRTC - Web Real Time Communications. It is an emerging standard that gives web browsers Real-Time Communications (RTC) capabilities via simple JavaScript APIs.