GLOSSARY

Business Communications A to Z

A HELPFUL COMPENDIUM OF UNIFIED COMMUNICATIONS & IP TELEPHONY TERMS, ACRONYMS AND PHRASES

Every industry has its communications short-cuts: those specialized phrases, abbreviations and insider terms that can fly by in a white paper or conversation as though the listener is already well-versed in the field. IP Telephony and Unified Communications is no exception. Today's sophisticated integration of business phones with UC applications offers a multitude of end-user features made possible by lightning-fast, low-cost Internet distribution, advanced communications protocols, ingenious device programming, and rich multimedia enhancements. It is a "must have" for all businesses, no matter the size. ShoreTel offers this glossary to take some of the complexity out of talking VoIP and UC.

ATM – Asynchronous Transfer Mode

ACD – Automatic call distribution

ANI – Automatic number identification

BYOD – Bring-your-own-device

A

Abandoned call – An abandoned call is when the caller disconnects while he or she is on hold and waiting for a call center agent. The caller may call back at a later time, but may become dissatisfied and call a competitor. As the call volume increases, the abandon rate usually increases. Experts consider the optimum abandoned call rate to be 3 percent to 5 percent.

Asynchronous Transfer Mode (ATM) – ATM is a type of wide-area network circuit that is used to transmit voice, video and data. All information sent over an ATM network is broken into packets of a fixed size, which means the performance is predictable and manageable.

Automated attendant – An automated attendant offers a customizable way for an organization to quickly route incoming calls to their destinations. Organizations can use automated attendants to connect callers with the right parties faster and hire fewer operators or receptionists.

Automatic call distribution (ACD) – An ACD is a system that receives incoming phone calls and routes them to the appropriate contact center agent. ACDs are used in businesses and contact centers that handle large volumes of incoming callers who may require assistance from any number of different people.

Automatic number identification (ANI) – ANI is a feature that allows the telephone company to capture the billing phone number of the calling party. ANI is similar to caller ID, but with ANI, the caller's phone number and line type are captured even if caller ID is blocked.

Availability – Availability of any IT system is based on the probably of a hardware failure. Availability takes into account the type and number of hardware components in a system and the mean time between failure (MTBF) for these components. If an IP telephony switch has a predicted MTBF of approximately 135,600 hours, and each failure requires one hour mean time to repair (MTTR), then availability is 99.9993%. This achieves five-nines of availability, and it means the switch will be unavailable for one hour every ten years.

B

Bring-your-own-device (BYOD) – With the huge popularity of mobile devices, many people want to use their personal smartphones, tablets and other mobile devices to access business applications and information. They won't need to carry different mobile devices for work and personal use, and they will be able to make and receive personal and business calls, while maintaining their business and personal personas. Companies can also save the cost of buying smartphones

and tablets for their workers. A successful BYOD initiative requires careful management and security to protect the business from risk.

C

Call center – A central location of an organization that handles high volumes of inbound and outbound calls, typically for sales, service or support.

Call detail record (CDR) – Business phone systems generate detailed logs of calls, which can be saved for processing and analysis. Service organizations that bill by the hour, such as legal and advertising firms, often input the information from call detail records into their customer billing systems.

Caller identification (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. Caller ID can be blocked by the calling party.

Capital expense (capex) – Capex includes all of the equipment required, including hardware and software, planning, installation, initial setup and integration with other systems. Capex declines over time, as IT products continue to improve in price/performance and new features. See operational expenses (opex).

Click to call – The ability to launch a phone call directly from the contact list on your computer.

Cloud-based VoIP or unified communications (UC) – Most IP voice and unified communications systems today are deployed on-premises, but cloud services are becoming increasingly popular. With a cloud service, costs are more predictable than with an on-premises solution. A cloud-based business phone system is both hosted and managed by the provider. The VoIP or unified communications system is located in the provider's secure data centers and the provider owns, configures and manages it.

Communications-enabled applications – These applications have directly integrated communications functionality. Key applications include collaboration, contact center, notification applications, reporting and analytics.

Communications-enabled business processes (CEBP) – When business applications are integrated with communications applications to improve operations and workflow, it is known as communications-enabled business processes. CEBP can be used in collaboration, contact center, notification applications, and reporting and analytics tools.

Computer telephony integration (CTI) – CTI refers to any technology that enables integrated interaction on a telephone and a computer. Examples include screen pops, automatic dialing and call center functions.

Conferencing – Conferencing allows two or more people to interact using audio (audio conferencing), audio and video (video conferencing), or a combination of audio, video and information sharing using a Web browser (web conferencing). Conferencing reduces the need for travel and encourages collaboration among employees, customers and partners who may be at different physical locations.

Contact center – A contact center handles high volumes of inbound and outbound calls, Web chats, instant messaging, emails, faxes and other forms of communications, typically for sales, service or support. A contact center also typically uses blended agents, who can manage multiple forms of voice and data-centric customer communications.

Customer relationship management (CRM) – Software that manages all aspects of an organization's interactions with its customers, clients and prospects.

D

Denial of Service (DoS) – DoS or distributed DoS (DDoS) attacks are favorites of criminals, political operatives as well as disgruntled customers, former employees and social protestors. DDoS attacks have become highly organized and complex, which makes them harder to fight. In a DDoS attack, the network is bombarded with traffic, resulting in a slowdown or crash of the applications and servers. Every organization using IP telephony should have backup telephone lines in case of a DDoS attack.

Direct inward dial (DID) – DID is used for call routing, and it enables external callers to contact a user directly at his or her unique phone number. In the UK, it is called direct dial-in (DDI).

E

Employee-liable mobile device – A mobile device, such as a smartphone or tablet, that's owned by the employee, not the organization, but is used to access corporate applications and data.

Enhanced 911 (E911) – When IP telephony or mobile phone users dial 911, E911 is used to allow the location of the user to be known to the call receiver. E911 support is required by the FCC for mobile phones and IP phones.

Ergonomics – The science of designing products, machines and systems to maximize the safety, comfort and efficiency of the people who use them. Ergonomics takes into

CDR – Call detail record

Caller ID – Caller identifiication

Capex – Capital expense

CEBP – Communications-enabled business processes

CTI – Computer telephony integration

CRM – Customer relationship management

DoS – Denial of service

DID – Direct inward dial

E911 – Enhanced 911

account psychology, physical measurement, environment and more to ensure that products are adapted to suit workers and their specific needs. Keep ergonomics in mind as you look at the handsets and software of unified communications systems.

F

Find me, follow me – This IP telephony feature allows an employee to program his or her extension to ring based on status, such as allowing the call through when he or she is in the office, forwarding to a mobile phone when there is no answer, or forwarding to a colleague when the line is busy. Find-me, follow-me 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 firewall sits between two networks, such as a company's network and the Internet, to prevent access by unauthorized users. A firewall monitors and controls the flow of data in and out of the network and can intercept, analyze and stop a wide range of malicious attacks. Make sure your company's firewall supports IP voice.

Fixed mobile convergence (FMC) – FMC solutions integrate cellular services with private communications networks, either wired or wireless. Users are accessible via a single number and can use of enterprise telephony features from their mobile phones. FMC can provide cost savings by eliminating expensive roaming charges on mobile networks, especially for international calls. See also mobile unified communications.

Frame relay – Frame relay is a type of wide-area network connection. Frame relay circuits are less expensive than private leased lines because the carrier shares the frame relay bandwidth among many customers. When properly engineered, frame relay can provide a cost-effective way to transmit IP telephony traffic and still guarantee quality of service.

G

G.711 – G.711 is an ITU-U standard that describes the 64-kbps pulse code modulation (PCM) voice coding technique. In G.711, encoded voice is already in the correct format for digital voice delivery in the public switched telephone network or through PBXs.

H

H.323 – H.323 is a standard that facilitates videoconferencing over packet-based networks. It includes a complete suite of protocols for audio, video and data

FMC – Fixed mobile coverage

IM – Instant messaging

IVR – Interactive voice response

conferencing. SIP and MGCP protocols have become more popular than H.323 over time.

Hosted VoIP – In a hosted VoIP or unified communications system, the carrier, vendor or solution provider runs and manages the system. Costs are more predictable than with the traditional on-premises model where businesses buy, build and manage their own phone systems.

Hunt groups – On a business phone systems, hunt groups ensure that all calls are answered by a live person, rather than having to wait in a queue or go to voicemail. An incoming call can ring extensions in a specified sequence or ring multiple extensions at once, depending on the company's preference.

I

IEEE 802.11 wireless LAN – Wireless LANs use radio technologies to provide connectivity for smartphones, tablets, IP phones, sensors and many other types of devices. 802.11 wireless LANs operate in the 2.4 GHz and 5 GHz radio bands. 802.11n is the newest and fastest technology and delivers up to 100Mbps performance–similar to or even better than wired networks. Older (but still compatible) technologies include 802.11a (54Mbps), 802.11b (11Mbps) and 802.11g (54Mbps).

IEEE 802.11e – 802.11e is an enhancement to the 802.11 wireless LAN specifications that provides quality-of-service features, including the prioritization of data, voice and video. To ensure a quality user experience, quality of service should be used for real-time applications such as voice and video.

IEEE 802.11i – 802.11i improves wireless authentication and encryption for wireless LANs. Together, 802.11i and Wi-Fi Protected Access 2 (WPA2), from the Wi-Fi Alliance, are used to secure wireless networks.

IEEE 802.1p – 802.1p is a specification that gives Ethernet switches the ability to identify and prioritize traffic into eight classes of service. 802.1p works at the media access control (MAC) layer, or Layer 2.

IEEE 802.1X – 802.1X authentication should be used to verify a user's identity on a wireless LAN, so that unauthorized guests are not allowed to use the network. Laptops, tablets and smartphones can support 802.1X authentication, but Wi-Fi IP phones, which have less computational capacity, use simpler methods such as MAC address authentication or username and password.

Instant messaging (IM) – Real-time communication over the Internet using text-based messages. Often a key component of UC.

Interactive voice response (IVR) – IVR is a method of communication between customers

and an automated system through the use of a telephone keypad or speech recognition.

IP contact center – Call centers using traditional PBXs can find that it is difficult and expensive to gradually increase the number of agents. With an IP-based contact center, companies can more easily growth their contact centers one agent at a time. In addition, IP contact centers can span across many locations, which gives the business greater flexibility than having to assemble a skilled team of agents in a single location. Agents can sign in from wherever they are (even at home) and be instantly online as part of the contact center team. Also called a virtual contact center or distributed contact center.

IP PBX – An IP PBX is a business phone system that delivers voice or video over a data network using IP. The previous generation of PBXs used time-division multiplexing, rather than IP.

IP phones – IP phones plug directly into the network and perform the necessary analog-to-digital and/or digital-to-analog conversions. You may need different models of IP phones based on various segments in your user population. For example, the legal department may need multi-line handsets with easy conference call capabilities, while the manufacturing floor needs a phone with fewer bells and whiles but good, loud sound and a rugged exterior.

IP telephony – IP telephony uses the IP network to carry voice, fax and other types of information that traditionally have been sent over the public switched telephone network. Also called voice over IP (VoIP).

IP telephony system architecture – Some IP telephony systems use a distributed system architecture, while others use a centralized architecture. In a centralized architecture, a central call control server provides dial tone for all phones. In a distributed model, the phone service is handled by multiple call control servers, which can be in different locations, and call control is provided by each switch in the system. A distributed system has greater reliability and resilience.

IP VaaS – IP voice as a service is a small but growing offering as a VoIP telephony solution. Rather than buying and managing the VoIP infrastructure internally, clients receive voice services via a third-party provider that hosts, delivers and manages the operation remotely. The infrastructure is typically shared with other clients. (see Hosted VoIP).

J

Jitter – Jitter can cause strange sounds to contaminate voice calls and users will complain about degraded voice quality. Jitter is caused by network congestion, how routers and switches queue up packets as well as network routing policies, such as traffic engineering or MPLS paths used by carriers. Latency, jitter and packet loss all impact voice quality.

L

Latency – Latency is the time it takes for a caller's voice to be transported – packetized, sent over the network, depacketized and replayed – to the other person. Geographical distance and lower-speed network connections can cause delays. If latency is too high, the natural conversation flow will be interrupted. Latency should not exceed 100 milliseconds one way for toll-quality voice. However, participants can still carry on a decent conversation if latency is up to 150 milliseconds. Latency, jitter and packet loss all impact voice quality.

Leased lines – Leased lines are the most private wide-area network (WAN) connection, and they are direct point-to-point connections between your locations. Leased lines can be used for data, voice and Internet services. They are also the easiest type of WAN circuit to ensure a high quality of service for voice and other applications.

Least-cost routing – Used in voice communication to select the outbound communications path based on lowest cost.

M

Managed IP telephony services – In a managed services model, the customer owns the IP PBX and related equipment, and it resides either at the customer's office or in the provider's network, while the carrier or value-added reseller provides management and maintenance for the phone system.

Mean time between failure (MTBR) – MTBF is the predicted time between failures of a system during operation. MTBF is calculated as the average time between failures of a system. The hard disk drives, fans and power supplies of a business phone system are the components that are most likely to fail. Look for a system with the fewest moving parts.

Mean time to repair (MTTR) – The time to repair a device. A four-hour MTTR is the industry standard, which can be a challenge for businesses that want to maintain five nines of availability for the IT systems. Redundant systems are usually added to ensure the appropriate levels of availability. Modular, distributed systems also make repair easy and results in a lower MTTR.

Media gateway control protocol (MGCP) – MGCP, also known as H.248 and Megaco, is an ITU-U standard for handling the signaling

and session management needed during a multimedia conference. MCGP defines the communication between a media gateway, which converts the data for use on a circuit-switched network, to the format required for a packet-switched network and the media gateway controller.

Metro Ethernet – A metropolitan area network that is based on Ethernet standards and is used to connect subscribers to the Internet or to their own offices in the region. 10-Gbps Ethernet services are increasingly popular for metro Ethernet.

Mobile device management (MDM) – MDM software secures, monitors and manages mobile devices, such as smartphones and tablets. MDM software usually distributes applications and enforces data and configuration settings for mobile devices, and includes the ability to lock or delete data from a lost or stolen mobile device. MDM can also be provided as a service.

Mobile unified communications (mobile UC) – A mobile UC solution includes common telephony functions, presence status, messaging, calendar, visual voicemail, videoconferencing and document sharing/collaboration from a smartphone or tablet.

Mobility Router – A mobility router is a network appliance that combines IP telephony, enterprise wireless LANs, carrier cellular networks and location technology to extend voice and unified communications to mobile devices. A 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 battery life.

Moves, adds and changes (MACs) – The administration necessary when users or network components are moved, added or changed on the network. When the moves, adds and changes for your phone system need to be done by a reseller or solution provider, rather than your own IT administrator or office manager, expenses can quickly add up.

Multiprotocol Label Switching (MPLS) – Many large enterprise and service provider networks use MPLS to create full mesh connectivity across the wide-area network without having to manage virtual circuits.

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N+1 redundancy – A business phone system with a distributed architecture that allows for use of N+1 redundancy will provide high levels of reliability. N+1 redundancy means that extra parts, such as voice switches, can be added to provide redundancy and there is no single point of failure. A 1:1 redundancy means that

all hardware must be doubled, which doubles the hardware cost.

Net Promoter Score (NPS) –SATMETRIX's Net Promoter Score® is a globally recognized metric used to measure customer loyalty. NPS is a measurement of an individual's willingness to recommend a company or product to others. A score of 50 or above is considered world class.

Network assessment – A network assessment provides a comprehensive performance assurance and real-time verification of unified communications performance right to the users' desktops. A network assessment will help scout out problems before IP voice or video is deployed. Any business deploying IP voice should first conduct a network assessment.

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Operational expenses (opex) – Opex is the cost involved once a given service is operational, including operational management, user education, support, troubleshooting and related labor-intensive services. Opex costs increase over time, as personnel costs rise. See capital expenses (capex).

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Packet loss – Packet loss results in a metallic sound or conversation dropouts in IP phone systems. Packet loss can be caused by network congestion, distance and poor line quality. Because VoIP is a real-time service, there's no way to recover lost packets. Even one or two percent packet drop results in noticeably degraded voice quality. Latency, jitter and packet loss all impact voice quality.

Power over Ethernet (PoE) – With PoE, electrical power and data are transmitted on Ethernet cabling. On today's networks, IP phones plug directly into the Ethernet network, rather than having a separate power source. The IEEE 802.3af Power over Ethernet standard can be used to power IP phones, video surveillance camera and other devices. The original IEEE 802.3af standard provides up to 15.4 watt of DC power, where the newer IEEE 802.3at (also known as PoE Plus) provides up to 25.5 watts of power.

Presence status – The ability to see a colleague's presence status, such as in the office, away, on the phone, or in a meeting. A user's presence status should change automatically if the person is on the phone or scheduled to be in a meeting. Presence services are expanding to include the presence and location information between multiple sources.

MDM – Mobile device management

MACs – Moves, adds and changes

MPLS – Multi label switching

NPS – Net promoter score

Opex – Operational expenses

PoE – Power over ethernet

Private branch exchange (PBX) – A PBX telephone system is based on time-division multiplexing (TDM) and they were traditionally deployed by organizations until the arrival of IP-based phone systems in the late 1990s.

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. The International Telecommunications Union (ITU) creates the standards that govern the technical operations of the PSTN.

Q

Quality of service (QoS) – There are a number of ways to identify and mark which applications and traffic should receive high priority transmission over the network. These methods include virtual LANs, DiffServ, Type of Service, IEEE 802.1p, IP address, and source and destination ports. Different QoS methods may be used at Layer 2 and Layer 3, and to ensure that quality of service can be enforced end-to-end across the network, you must ensure that these QoS methods can be translated from one layer to another.

R

Real-Time Transport Protocol (RTP) – An IP standard that provides end-to-end network transport functions that can be used by application transmitting real-time voice, video or data over unicast or multicast network services.

Reliability – Reliability generally refers to hardware reliability. Reliability is determined by calculating how often the system fails compared to the percentage of the time that the system is available. In the telephony world, five nines of reliability is the acceptable benchmark. This means the system is available at least 99.999 percent of the time. Reliability is different than availability.

Return on investment (ROI) – ROI is used to justify a purchase decision. In IT, ROI is about the productivity an organization gains by implementing a particular technology. For unified communications, this productivity could be qualitative, such as customer satisfaction, or end-user productivity, such as increased call volumes or the productivity of contact center agents. ROI and total cost of ownership (TCO) are often confused. See also total cost of ownership.

S

SaaS – Software as a service, sometimes referred to as "on-demand software", is a virtual application service in which software applications and their associated data are centrally hosted on the cloud. The services provided are typically accessed by users via a web browser.

Seamless roaming – When a person walks from one location to another using a mobile device, he or she counts on roaming capabilities of the wireless LAN to keep the application or voice call connected. Invisible to the user, the underlying wireless infrastructure hands off the user from one wireless access point to the next, while performing the necessary re-association and re-authentication in the background.

Service level agreement (SLA) – An SLA is part of a service contract where the level of service is defined. SLAs can be between service providers and customers, or businesses and their internal departments. An SLA may describe the service uptime, application response time, application performance, time to repair, and other measurable details. An SLA for voice quality might include call completion rate, delay from when the last digit is dialed until a user hears a ringing or busy signal, and a mean opinion score to measure voice quality.

Session border controller (SBC) – If you are using SIP trunks to connect your company's different locations, you should add a session border controller to protect against malicious attacks. An SBC ensures that only approved traffic passes 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 signaling protocol used to establish a session in an IP network, from a simple telephone call to a multimedia conference session with many participants. SIP establishes, manipulates and tears down sessions. However, there are many different "flavors" of SIP, which can create interoperability issues. SIP is RFC 3261 from the Internet Engineering Task Force (IETF).

SI – Systems Integrators specialize in bringing together component subsystems into a whole andensuring that those subsystems function together, a practice known as system integration.

Skills-based routing – In a contact center, skills-based routing is used to automatically route calls to the most appropriate agent, based on criteria such as language, experience or technical expertise.

Softphone – A softphone, which delivers telephony capabilities to any PC or laptop, is particularly convenient for workers who travel frequently. With calls directed to a laptop and a headset plugged into the USB port, an employee can work from anywhere using the computer and its built-in microphone.

PBX – Private branch exchange

PSTN – Public switched telephone network

QoS – Quality of service

RTP – Real-time transport protocol

ROI – Return on investment

SLA – Service level agreement

SBC – Session border control

SIP – Session initiation protocol

T

TDM – Time-division multiplexing (TDM) is a type of signal multiplexing in which two or more bit streams share allocated space on one communication channel by physically taking turns on the channel via synchronized and error-corrected time slots of fixed length. TDM exponentially increases the amount of data that can be transmitted simultaneously.

Telephony denial of service (TDoS) – TDoS refers to automatically generated call floods received by businesses – usually their contact centers. For example, attackers can flood the contact center's IVR or agents with hundreds or thousands of concurrent calls to disrupt the business or to generate calls for revenue. See also denial of service.

Telepresence – Telepresence is a set of technologies that provides video and sound quality that is as good as being there. Telepresence has extended the capabilities of videoconferencing beyond the boardroom to tablets and other mobile devices, facilitating collaboration.

Throughput – How much bandwidth your system needs, which depends on the number of simultaneous calls, the voice encoding scheme used by the IP phone or softphone, and the signaling overhead.

Toll fraud – Insider abuse or a compromised system may allow outsider access to the phone system to make calls, often internationally. Toll fraud can cost companies millions of dollars.

Total cost of ownership (TCO) – TCO helps you determine which specific vendor's product you should purchase. TCO is the measure of what technology costs to own. For unified communications, these costs could include capital expenses such as hardware and software, implementation and training, system management, and network and long distance charges. The focus for organizations should be on choosing products that reduce these tangible costs over the lifetime of owning the system. TCO and return on investment (ROI) are often confused. See also return on investment (ROI).

Trunk line – Trunk lines connect the PBX to public switched telephone network. In older PBX phone systems, ISDN, T1, E1 and DS3 were commonly used for trunk lines. Today, SIP trunks are often used.

U

UCaaS – Unified communications as a service is a turn-key business communications solution whereby clients pay for shared use of an IP UC infrastructure which is owned and managed offsite by a provider (see Hosted VoIP).

UCaaS a more sophisticated communications solution than simple voice, and should include instant messaging, conferencing, presence and voicemail features, as well as advanced integration with email and other business process software, to be considereda truly unified communications solution.

Unified communications (UC) – UC provides a single interface through which users can access a wide range of communications options, including voice, instant messaging, email, conferencing and more.

Unified communications and collaboration (UCC) – UCC integrates multiple communications channels and enables communication and collaboration within workgroups and across company boundaries. UCC includes voice and telephony, conferencing, messaging, presence status and instant messaging, and these capabilities can be accessed from a variety of devices. Products may be a standalone suite or a portfolio of integrated applications and platforms.

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

V

VaaS – Voice as a Service. (See IP VaaS)

VAR – A value-added reseller is a company that adds features or services to an existing product, then resells it as an integrated product or complete "turn-key" solution. The added value can come from professional services such as integrating, customizing, consulting, training and implementation. Customers purchase systems from a reseller if they lack the time or experience to assemble the system themselves. Some OEMs chose to market their products and services solely through VARs.

Virtual contact center – A virtual contact center leverages an IP-based contact center solution that allows agents to work from different locations and even from home. A virtual contact center scales more effectively and allows businesses to leverage expertise across locations, rather than hoping to find a highly skilled team in one location. Also called an IP contact center.

Virtual LANs (VLANs) – Many organizations use virtual LANs (VLANs) to logically separate devices and departments on the same physical network. Isolating different groups' traffic from one another provides an additional layer of security. Giving voice traffic its own VLAN

will deliver a better user experience. Using VLANs also reduces the impact of multicast or broadcast traffic.

Virtual private network (VPN) – A VPN is a private network used by a company, and in many cases, its partners or associates, to communicate over a non-private network. VPN traffic can be carried over a public network infrastructure such as the Internet. Internet-based VPNs offer the least amount of administrative control to regulate and guarantee quality of service to ensure a good voice experience.

Virtualization – Virtualization is the creation of a virtual (rather than an actual) version of something, such as an operating system, server, storage device or network. For example, server virtualization essentially tricks the operating system into thinking that a group of servers is a single pool of computing resources, and that allows IT to run multiple operating systems simultaneously on the same machine. Virtualization allows companies to consolidate servers, increase operational flexibility and deliver higher application availability.

Voice over IP (VoIP) – VoIP is a method of sending voice calls over the IP data network. This can apply to the public Internet or private IP networks. Also called IP telephony and IP voice.

Voice over wireless LAN (VoWLAN) – Running IP voice over a wireless LAN.

W

WAN optimization – WAN optimization products accelerate your applications by eliminating redundant transmissions, staging data in local caching, compressing and prioritizing data, and streamlining chatty protocols. This makes more room for your real-time applications, such as voice and video.

WebRTC – Web Real-Time Communication) is an API definition being drafted by the World Wide Web Consortium (W3C) to enable browser to browser applications for voice calling, video chat and P2P file sharing without plugins.

Weighted Fair Queuing (WFQ) – WFQ is a quality of service method that is used on routers and switches. WFQ allows traffic flows to share the network, but provides prioritization for small, time-sensitive traffic. This way, a large traffic flow, such as a backup, will not clog the pipe or create lengthy delays for smaller flows, such as voice.

Wide area network (WAN) – A telecom network that spans great distances, often across a metropolitan or regional area, but also across countries.

Wideband audio –Wideband audio (50 Hz to 7,000 Hz) is preferred for business phone systems, because it makes conversations sound better and reduces errors in translation. Wideband audio more than doubles narrowband (300 Hz to 3,400 Hz). An IP PBX should support both wideband and narrowband.

Wi-Fi – Wi-Fi is a trademark of the Wi-Fi Alliance, an industry organization. Wi-Fi is used to describe products that have passed the organization's interoperability tests; however, the term Wi-Fi is commonly used to describe wireless LANs.

Wireless LAN – Wireless LANs use radio technologies to provide connectivity for smartphones, tablets, IP phones, sensors and many other types of devices. Smaller wireless LANs use access points (APs) to provide the service, while larger wireless LANs also use mobility controllers to control and manage the many access points.

Wireless Multimedia (WMM) – WMM is used to deliver quality of service for voice and other applications on wireless LANs. WMM defines four priority levels to support different types of traffic, including voice, video, best effort for data and background traffic. WMM is a Wi-Fi Alliance certification that's based on the IEEE 802.11e standard.

VPN – Virtual private newtork

VoIP – Voice over IP

VoWLAN – Voice over wireless LAN

WebRTC – Web real-time communications

WFQ – Weighted fair queuing

WAN – Wide area newtork

WMM – Wireless multimedia

About ShoreTel

ShoreTel is a provider of business communication solutions whose brilliantly simple unified communications platforms, applications and mobile UC solutions promise a new rhythm of workforce engagement and collaboration. With costly complexity eliminated by design from its award winning, all-in-one IP phone system, UC and contact center solution, and its industry leading hosted business phone system, workers enjoy a freedom and self-reliance that other providers can't match. Users have full control to engage and collaborate, no matter the time, place or device, for the lowest cost and demand on IT resources in the industry. ShoreTel is headquartered in Sunnyvale, California, and has regional offices and partners worldwide. For more information, visit shoretel.com or shoretelsky.com

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