

**ACD:** Average Call Duration is the average duration of the calls routed by a VoIP provider. It is a quality parameter given by the VoIP providers.

**API:** An application programming interface (API) is a source code interface that a computer system or program library provides to support requests for services to be made of it by a computer program. We currently offer an XML API to query our current rates.

**ASR:** Answer Seizure ratio (ASR) is the number of successfully answered calls divided by the total number of calls attempted (seizures). Since busy signals, calls not answered and other rejections by the called number count as call failures, the calculated ASR value can vary depending on user behaviour.

**ATA:** An analog telephony adapter, or analog telephone adapter, (ATA) 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.

**CDR:** Telephone exchanges generate so called Call Detail Records (CDRs) which contain detailed information about calls originating from, terminating at or passing through the exchange. Not surprisingly CDRs are used for billing.

**CNAM:** A CNAM database contains calling party names to be used when identifying a calling party. We offer optional CNAM service on our US and Canadian DIDs as well as on our toll free dids.

**DHCP:** Dynamic Host Configuration Protocol is a client/server protocol that automatically provides an Internet Protocol (IP) host with its IP address and other related configuration information such as the subnet mask and default gateway

**DID:** Direct Inward Dialing. "DID" numbers have particular relevance for VoIP communications. In order for people connected to the traditional PSTN network to call people connected to VoIP networks, DID numbers from the PSTN network are obtained by the administrators of the VoIP network, and assigned to a gateway in the VoIP network. The gateway will then route calls incoming from the PSTN across the IP network to the appropriate VoIP user. Similarly, calls originating in the VoIP network will appear to users on the PSTN as originating from one of the assigned DID numbers, if the user setup his callerid accordingly.

**FTP:** File Transfer Protocol is a standard network protocol used to transfer computer files between a client and server on a computer network.

**IAX:** IAX is the Inter-Asterisk eXchange protocol used by Asterisk, a dual licensed open source and commercial PBX server from Digium and other softswitches and PBXs. It is used

to enable VoIP connections between servers, and between servers and clients that also use the IAX protocol.

**IP:** Internet Protocol is the method or protocol by which data is sent from one computer to another on the Internet. Each computer (known as a host) on the Internet has at least one **IP** address that uniquely identifies it from all other computers on the Internet.

**IP PBX:** Internet Protocol- based Private Branch Exchange

**MGCP:** Media Gateway Control Protocol is a signaling and call control communications protocol used in Voice over IP (VoIP) systems.

**NAT:** Network Address Translation is a methodology of remapping one IP address space into another by modifying network address information in Internet Protocol (IP) datagram packet headers while they are in transit across a traffic routing device.

**PBX:** Private Branch Exchange is a private telephone network used within a company. Users of the **PBX** phone system share a number of outside lines for making external phone calls.

**PSTN:** Public Switched Telephone Network is the world's collection of interconnected voice-oriented public telephone networks, both commercial and government-owned. It's also referred to as the Plain Old Telephone Service (POTS)

**QoS:** Quality of Service is the idea that transmission rates, error rates, and other characteristics can be measured, improved, and, to some extent, guaranteed in advance.

**RAS:** Registration, Admission, and Status protocol is a management protocol between endpoints (terminals and gateways) and gatekeepers. The RAS is used to perform registration, admission control, bandwidth changes, status, and disengage procedures between endpoints and gatekeepers. Registered memory.

**RTP:** Real-time Transport Protocol is a network protocol for delivering audio and video over IP networks. RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications, television services and web-based push-to-talk features.

**SCP:** Secure Copy Protocol is a network protocol, based on the BSD RCP protocol, which supports file transfers between hosts on a network. SCP uses Secure Shell (SSH) for data transfer and uses the same mechanisms for authentication, thereby ensuring the authenticity and confidentiality of the data in transit.

**SER:** SIP Express Router (SER) is a high-performance, configurable, free software SIP (cit. RFC 3261 ) server . It can act as SIP registrar, proxy or redirect server. SER features

presence support, RADIUS/syslog accounting and authorization, XML-RPC-based remote control, etc. Web-based user provisioning, serweb, is available. SER's performance allows it to deal with operational burdens, such as broken network components, attacks, power-up reboots and a rapidly growing user population. SER can be configured for many scenarios including small-office use, enterprise PBX replacements and carrier services. SER is publicly available under the terms of the GNU General Public License.

**SIP:** Session Initiation Protocol is an Internet Engineering Task Force ([IETF](#)) standard protocol for initiating an interactive user session that involves multimedia elements such as video, voice, chat, gaming, and virtual reality.

**VoIP:** Voice over Internet Protocol The transmission of voice over the Internet as digital packets rather than the traditional circuit-committed protocols of the PSTN. VoIP uses real-time protocol (RTP) to help ensure that the packets get delivered in a timely way.

**CallerID:** Caller ID (caller identification or CID, and more properly calling number identification - CNID) is a telephony service that transmits the caller's telephone number to the called party's telephone equipment during the ringing signal or when the call is being set up but before the call is answered. Where available, Caller ID can also provide a name associated with the calling telephone number. The information made available to the called party is visible on a small liquid crystal display imbedded on the telephone, or on a separate unit which is connected to the telephone.

**Call Origination:** Call Origination, also known as voice origination, refers to the collecting of the calls initiated by a calling party on a telephone exchange of PSTN, and handing off the calls to a VoIP endpoint or to another exchange or telephone company for completion to a called party.

**Codec:** Voice transmission is analogical, whereas the data network is digital. The process to sample analogical waves into digital information is made by an encoder-decoder (CODEC). There are many standards to sample an analogical voice signal into a digital one. The process is often quite complex. Most of the conversions use pulse code modulation (PCM) or variations.

**Inbound:** We refer to inbound as traffic we receive and that is directed to you, such as calls to your DID numbers.

**Jitter:** Jitter is a typical problem of the connectionless networks or packet switched networks. Due to the information is divided into packets each packet can travel by a different path from the emitter to the receiver. Jitter is technically the measure of the variability over time of the latency across a network. Real time communications (for example VoIP) usually have quality problems due to this effect. In general, it is a problem in

slow-speed links or with congestion. It is hoped that the increase of QoS (quality of the service) mechanisms like priority buffers, bandwidth reservation or high-speed connections (100Mb Ethernet, E3/T3, SDH) can reduce jitter problem in the future although it will keep on being a problem for a long time.

**Latency:** Latency has the reputation of being the enemy of VoIP. It is also called lag. Latency is the time between the moment a voice packet is transmitted and the moment it reaches its destination. It of course leads to delay and finally to echo. It is caused by slow network links. This is what leads to echo. There are two ways latency is measured: one direction and round trip. One direction latency is the time taken for the packet to travel one way from the source to the destination. Round-trip latency is the time taken for the packet to travel to and from the destination, back to the source. In fact, it is not the same packet that travels back, but an acknowledgement. Latency is measured in milliseconds (ms) - thousandths of seconds.

**Packet Loss:** VOIP is not tolerant of packet loss. Even 1% packet loss can "significantly degrade" a VOIP call using a G.711 codec and other more compressing codecs can tolerate even less packet loss. (Intel whitepaper)

**Prepaid Model:** Prepaid refers to services paid for in advance. Examples include tolls, pay as you go cell phones, and stored-value cards such as gift cards and preloaded credit cards. Prepaid options can have substantial cost reductions over postpaid counterparts because they allow customers to monitor and budget usage in advance.

**Predictive Dialer:** A predictive dialer is a computerized system that automatically dials batches of telephone numbers for connection to agents assigned to sales or other campaigns. Predictive dialers are widely used in call centers.

**Soft Switch:** A soft switch is a central device in a telephone network which connects calls from one phone line to another, entirely by means of software running on a computer system. This work was formerly carried out by hardware, with physical switchboards to route the calls.

**Sub-Account:** A sub-account can be used to break down an account into multiple smaller accounts. This could be for better tracking of detailed budgets and expenses, connect multiple PBX, connect devices from different locations or simply to connect multiple devices to our service without the need of a PBX. With the use of sub accounts, the customer doesn't need to open multiple accounts and manage multiple balances to use the service.

**Termination:** Call Termination, also known as voice termination, refers to the handing off or routing of calls from one telephone company, also known as a carrier or provider, to

another telephone company. The terminating point is end point. The originating point is the party who initiates the call. This term is highly used when referring to calls while using voip: a call initiated as a VoIP call is terminated using the PSTN The opposite of call termination is call origination, where a call initiated from the PSTN is terminated using VoIP.