3Com® IP Telephony Acronyms and Glossary

TECHNOLOGY GUIDE

ACRONYMS

AA
Auto-Attendant

ACD
Automatic Call Distribution

ADSL
Asymmetric Digital Subscriber Line

AES
Advanced Encryption Standard

ANI
Automatic Number Identification

ANSI
American National Standards Institute (www.ansi.org)

API
Application Programming Interface

ATM
Asynchronous Transfer Mode

B2BUA
Back-to-Back User Agent

BRI
Basic Rate ISDN

CAS
Complement Attendant Software

CDR
Call Detail Reporting

CLI/CLID
Calling Line Identification

CO
Central Office

CoS
Class of Service

CRM
Customer Relationship Management

CTI
Computer Telephony Integration

DCA
Desktop Call Assistant

DES
Data Encryption Standard

DHCP
Dynamic Host Configuration Protocol

DID/DDI
Direct Inward Dialing/Direct Dialing Inward

DiffServ
Differentiated Services

DNIS
Dialed Number Identification Service

DSS/BLF
Direct Station Selection/Busy Lamp Field

DTMF
Dual-Tone MultiFrequency

E911
Enhanced 911

E&M
Earth and Magnet (signaling)

ETSI
European Telecommunications Standards Institute (www.etsi.org).

FXO
Foreign eXchange Office

FXS
Foreign eXchange Subscriber

HTTP
Hypertext Transfer Protocol

IEEE
Institute of Electrical and Electronic Engineering (www.ieee.org)

IETF
Internet Engineering Task Force (www.ietf.org)

IM
Instant Messaging

IMAP
Internet Message Access Protocol

IP
Internet Protocol

IPS
Intrusion Prevention System

IPsec
IP Security

ISDN
Integrated Services Digital Network

ISO
International Organization of Standardization (www.iso.org)

ITU
International Telecommunication Union (www.itu.int)

IVR
Interactive Voice Response

Kbps or Kb/s
Kilobits per second

KTS
Key Telephone System

LAN
Local Area Network

LCR
Least Cost Routing

Mbps or Mb/s
Megabits per second

MIB
Management Information Database

Modem
Modulator/Demodulator

MOS
Mean Opinion Score

MPLS
Multi-Protocol Label Switching

NAT
Network Address Translations

NBX®
Network Branch eXchange

NCP
Network Call Processor

NOS
Network Operating System

NPA
Numbering Plan Area

PABX
Private Area Branch Exchange

PAN
Personal Area Network

PBX
Private Branch Exchange

PCI
Peripheral Component Interconnect

POP3
Post Office Protocol 3

ACRONYMS—continued

POTS
Plain Old Telephone Service

PRI
Primary Rate Interface

PSTN
Public Switched Telephone Network

QoS
Quality of Service

RSVP
Re**s**er**v**ation **P**rotocol

RTP
Real-Time Protocol

RTSP
Real-Time Streaming Protocol

SDP
Session Description Protocol

SIP
Session Initiation Protocol

SIMPLE

SIP Instant Messaging and Presence Leveraging Extensions

SMDI
Simplified Message Desk Interface

SMDR
Station Message Detail Recording

SMTP
Simple Mail Transfer Protocol

SNMP
Simple Network Management Protocol

SRTP
Secure Real-Time Transport Protocol

TAPI
Telephony Application Programming Interface

TCP/IP
Transmission Control Protocol/ Internet Protocol

TLS
Transport Layer Security

TSAPI
Telephony Server Application Programming Interface

UCD
Uniform Call Distribution

UDP
User Datagram Protocol

UMTS
Universal Mobile Telecommunications System (www.umts-forum.org)

VCX™
Voice Core eXchange

VLAN
Virtual LAN

VM/MS
Voicemail/Messaging System

VoIP
Voice over Internet Protocol

VPIM
Voice Profile for Internet Mail

VTL
Virtual Tie Line

W3C
World Wide Web Consortium (www.w3.org)

WAN
Wide Area Network

Wi-Fi
Wireless Fidelity

WLAN
Wireless LAN

XML
Extensible Markup Language

TERMS 3—C

3G
Third-generation digital cellular mobile standard for delivering high-speed data to mobile devices; will eventually deliver packet-based mobile communications as IP Multimedia Subsystem (IMS) architectures are implemented.

802.11a/b/g
Standards for wireless networking using unlicensed radio spectrum for Wi-Fi communications.

Abbreviated Dialing
Also known as speed dialing, allows a list of frequently called numbers to be stored in system memory and accessed via a two- or three-digit code.

Account Codes
Allows a phone user to enter an account code that will appear in a record of calls made or received; codes can be forced or unforced; optionally, they can be verified against an allowed code list defined by an administrator.

Ad-hoc Conference
Conference bridge facility where parties can call a number at any-time and be placed in conference, sharing a pool of conference ports with no guarantee of availability; unlike scheduled conferences with pre-booked resources; also known at meet-me conferences.

Attendant Console
A standard telephony device that shows the status of each extension in a telephone system, allowing a receptionist to connect incoming calls to the requested destination.

Authorization Codes (CoS Override)
Allows phone users to temporarily override the configured Class of Service (CoS) on a phone and access additional services by using their authorization code; usually enabled by entering a user's extension number and PIN.

Auto-Attendant (AA)
An application that answers incoming calls with a menu of options; a recorded greeting may be followed by a transfer to a live attendant or station, or it may provide dialing instructions for reaching certain users, departments or a live attendant.

Auto-Discovery
A feature that "discovers" a new telephone or other device on the network and provides it with service without specific directions from a system administrator.

Automatic Call Distribution (ACD)
A call management system that routes calls to queues so they can be answered by appropriately skilled agents. The ACD can manage individual and groups of ACD agents, as well as recorded announcements presented to callers; the software also can provide database reports on both calls and agents.

Automatic Hold Recall
Allows calls to be placed on hold and recalls the called party if there is no response to the call within a specified period of time.

Auto-Redial
A modem, fax or telephone feature that redials a busy number a fixed number of times before disconnecting.

Auto-Relocation
A feature that allows a telephone to keep its extension number and personal and systems settings when it is connected to a different port on the network.

Back-to-Back User Agent (B2BUA)
A device that follows the rules of the Session Initiation Protocol (SIP)—standard RFC3261—for a user agent (client), but relays signaling to other clients or proxies; enables comprehensive features to be delivered to SIP clients without the constraints of being a SIP proxy.

Basic Rate ISDN (BRI)
Two-channel ISDN service usable for voice or data (2 x 64 Kbps).

Bit
Unit of information within binary data represented by one or zero.

Bluetooth
Wireless standard for short-range, device-to-device communications (www.bluetooth.com).

Bridged Appearance
Allows joint management of an extension by enabling specified phones to simultaneously share a line appearance; typically used in an executive/admin environment where the admin can see, answer and manage calls made to multiple telephones.

Buffer
Area in a device for temporary storage of data in transit; can accommodate differences in processing speeds to facilitate a clear transmission between devices by storing data blocks until they are ready to be processed by a slower device.

Busy Override
Allows users to override a busy signal to break into an existing conversation; most systems, but not all, offer this feature with a warning tone that is given to all parties of a current conversation.

Call Accounting System
Usually an external application to a telephony system which merges call detail records with call costs, facilitating client and/or departmental billing and helping verify telephone company bills.

Call Coverage
Pre-configured routing for calls which are unanswered by a called extension or call forwarding rules; frequently configured to voice mail.

Call Detail Record (CDR)
Saved call information relating to number dialed, duration, etc..

Caller Identification
- CLID (Calling Line IDentification): indicator on the phone receiving a call the caller's name or phone number; Caller ID is a service that a local phone company or service provider can offer that displays a CLID.
- ANI (Automatic Number Identification): indicator of caller data displayed on the phone receiving the call. When the call is routed to a PBX via an ISDN trunk from the public telephone network, the caller's phone

number (ANI) is automatically processed for future reference.

- DNIS (Dialed Number Identification Service): indicator of data about the number called that is displayed on the caller's phone. When the call is routed to it, the PBX processes the DNIS for display.

Call Forward
Enables users to arrange for calls to be sent to another phone in defined situations; can be implemented for:

- busy calls only
- no-answer calls
- all calls
- fixed calls: forwards to an assigned location (generally customizable), such as a manager to an admin
- override/return calls: allows a call to override a forwarded call and return it to the call forwarder's phone
- off premise calls: allows calls to be forwarded off-site

Call Park
Allows users to transfer calls to Parking or Orbit zones where the calls remain on hold until picked up at the same station or at another location.

Call Park (with Recall)
An automatic function of some systems that recalls either the original phone or the attendant to ensure that the caller is not forgotten when a call is placed in a Park location for an extended period of time.

Call Pickup (Directed, Group)
Enables any user or a group of users to pick up a call ringing at an unattended station; one group member can dial an access code to pick up a call ringing at an unattended member's station.

Call Waiting
An indication to a user who is engaged on one call that another is trying to connect; a recorded message can be used to let the caller know that the called user is being alerted.

Camp-On Busy
Allows a caller to request an automatic callback from a busy extension when that user is free.

Central Office (CO)
Term used by service providers for the site that houses the switching equipment for their public telephony services.

Circuit Switching
A bandwidth allocation method for using a fixed portion of the network and creating end-to-end connections that are open during the full duration of the communications session.

Class of Service (CoS)
User profiles defined by a system administrator to restrict or allow access to specific features and calling privileges and services.

Codec
Coder/decoder for handling audio signals over a data network; e.g., G.711 and G.729.

Complement Attendant Software (CAS)
A desktop call control application with advanced features to improve call handling for receptionists.

Computer Telephony Integration (CTI)
The integration of business applications with voice systems through specialized software applications (middleware); applications include contact center software which will manage call routing based on business rules and enable viewing caller information on a PC when a call is received.

Conference
Pre-scheduled or ad-hoc calls with multiple parties.

Desktop Call Assistant (DCA)
On-screen dialing and contact management software.

Direct Group Calling
Allows users to call a group of stations such as a sales department by dialing a group code number.

Direct Inward Dial (DID)/Direct Dialing Inwards (DDI)
Connects calls from the public network directly to the dialed extension without attendant assistance, allowing easier and faster access to called parties.

Directory Service (on LCD phones)
Look-up functionality that allows users to enter a name or set of characters on a phone's keypad to display a needed phone number.

Distinctive Ringing
A system configuration that identifies callers or their locations by unique ringing patterns; may also be combined with Personal Ringing.

Do Not Disturb
A feature that prohibits calls from terminating at a particular phone.

Do Not Disturb with Override
A capability that allows authorized users to override the Do Not Disturb feature and have their calls ring the phone that is in Do Not Disturb mode.

Door Phone
An intercom box or a combination intercom box/electronic door lock that can be installed on an outside door and directly connected to a specific phone or to the system.

Dual-Tone MultiFrequency (DTMF) Signaling
A tone-based signaling system that can be used for initial dialing or leveraged to communicate to a call processing application after the initial call is established. Interactive voice response (IVR) systems may prompt the user to enter information via the dial pad that will be sent via DTMF tones that correspond to the key(s) that were pressed.

Dynamic Host Configuration Protocol (DHCP)
Software that dynamically assigns IP addresses to network devices such as workstations, IP phones and gateways, allowing network devices to be easily moved from one subnet to another without administrative attention.

E1
European digital transmission format that carries signals at 2 Mbps (32 channels at 64 Kbps, with 2 channels reserved for signaling and controlling).

E—I

E&M (Earth & Magnet) Signaling
A legacy signaling system used over two- and four-wire analog connections.

Enhanced 911 (E911)
An emergency system, based on database information, that transmits a caller's phone number and location directly to emergency responders.

Ethernet
The most common local area network (LAN) technology, defined by the IEEE 802.3 standard, that lets devices contend for access to a shared network.

Executive Override of Privacy
Allows authorized users to override the privacy feature and join an existing two-party conversation.

Extensible Markup Language (XML)
A standard for storing and transmitting data between systems; allows data to be simply structured and shared between databases and applications.

External Alerting Device
External bells, lights or gongs that can be placed in special locations to facilitate the handling of calls.

External Page Interface
Connections for an external loud speaker able to page one or more zones.

Fax (Facsimile)
Transmission of a document over telecommunications channels using a modem or ISDN.

Fax Machine
A device that converts a document to a digital form suitable for sending and receiving over telecommunications systems and that allows the printing of a "fax."

Flexible Station Numbering
Allows users to customize their own phone numbering scheme and feature and trunk access codes.

Foreign Exchange Office (FXO)
An interface that connects local calls to a PSTN central office or to a PBX that does not support E&M signaling.

Foreign Exchange Station (FXS)
An interface that connects directly to a standard analog phone, fax machine or similar device, supplying ringing voltage, dial tone and similar signals to the endpoint device.

Frame Relay
A high-speed packet switching protocol in which unprocessed packets are relayed from a switch's input port to output port: used for wide area network (WAN) connectivity.

Gateway
A device that interconnects networks with different, incompatible communications protocols; specifically used to connect VoIP networks with analog phones, PBXs and PSTN trunks.

H.323
A comprehensive standard of the International Telecommunication Union (ITU) for realtime voice and videoconferencing (including application sharing and whiteboarding) over packet networks—LANs, WANs and the Internet.

Hands-Free
A phone feature that allows users to answer incoming calls using an integral speaker and microphone without having to touch the phone. After a warning tone the phone automatically goes off-hook.

Hold (System, Exclusive)
Allows users to put callers on hold so that they can answer or initiate another call; calls can be placed on System Hold so that any other station user with the same line appearance can pick up the call, or they can be placed on Exclusive Hold so that only the station user who put the caller on hold can access the call.

Hot Lines (or Manual Lines/House Phones)
Allows a direct connection to be automatically established to another phone in the system when a user goes off-hook.

Hunt Groups
A group of extensions accessed via a single number and allowing calls to be distributed to multiple users, any of whom can answer the calls. Users can usually log in and log out of hunts groups. Administrators can determine behavior with a variety of hunt group types, including:

- linear: starts with the same extension in the group each time and continues ringing all the phones in the group until a user answers, or the last phone has been rung or a pre-determined time is reached
- circular: starts with ringing the next number in the group for each new call and continues through the extensions in the group until the call is answered or a pre-determined time is reached
- longest idle: rings the phone in the group that has been without a call for the longest time; continues to the next shortest idle until answered or a pre-determined time is reached

Hybrid Mode
A PBX operating mode in which some outside lines are grouped together in pools while other lines are assigned directly to buttons on telephones; users access outside lines by dialing a pool access code (see key mode).

Hybrid PBX
A PBX that is able to support both traditional analog or digital phones as well as IP phones.

I-Hold Indication
Provides multiline station users with an indication as to which holding line is the one that they placed on hold.

Internet Engineering Task Force (IETF)
Organization responsible for the many Internet standards such as HTTP, SMTP and SIP.

Internet Message Access Protocol (IMAP4)
A standard interface between an email client and an email server that displays headers with to/from addresses and subject (rather than entire messages) so that users can selectively open and download their messages.

I—N

Incoming Call Group
Allows central office trunk lines to be assigned to ring directly into a group of stations without being routed through the attendant.

Interactive Voice Response (IVR)
An automated system that can interact with a caller, providing additional services such as directory lookup or call transfers by collecting information through DTMF or speech recognition; often used in front of contact centers to collect customer specific information before connecting the caller to an agent.

Internal Paging
Allows users to place page calls to certain groups or all users on the system via the telephone speaker.

Intrusion Prevention System (IPS)
An inline security system for detecting and eliminating network-borne application or infrastructure attacks; uses deep packet inspection to identify and block transmission of malicious software such as worms, viruses and trojan horses.

IP Telephony
Voice services delivered over an IP network.

Integrated Services Digital Network (ISDN)
An international standard for switched, digital voice and data services based on 64 Kbps circuits.

I-Use Indication
Provides multiline station users with an indication of the line that is currently in use.

Jitter
The variation in latency (delay) for different packets to travel over a network; for real-time data such as voice transmission, jitter must be kept to a minimum.

Key Mode
A telephone system operating model in which each telephone in the system has buttons for each available outside line; also known as a square plan or a direct system inward access (DISA) system (see hybrid mode).

Key Telephone System (KTS)
A standalone, inhouse telephony system (often with relatively few users) in which all lines are directly connected to the telephone company central office, allowing users to place outside calls without dialing "9" first; connected phones have buttons for contacting on-premises phones as well as for making outside calls.

LAN Telephony
The convergence of voice and data traffic over a local area network.

Latency
The sum of all the delays in an end-to-end connection on a network transmission.

Least Cost Routing (LCR)
A call routing plan, similar to Auto Route Selection (ARS) that uses time-of-day, day-of-week and digit translation information to find the least expensive routes for long-distance calls.

Local Loop
The analog component of the PSTN that connects subscribers to a service.

Main/Satellite Service
Interconnectivity among multiple systems—such as those deployed in a business or school campus environment—provided by special trunks that allow almost transparent feature sharing with common numbering plans and access codes.

Manual Signaling
Allows users to send a manual signal or intercom tone to another phone by pressing a button.

Meet-me Conference
Conference bridge facility where parties can call a number at any-time and be placed in conference, sharing a pool of conference ports with no guarantee of availability; unlike scheduled conferences with pre-booked resources; also known as ad-hoc conferences.

Mean Opinion Score (MOS)
A commonly used measurement of voice quality derived from multiple callers rating call quality, thereby creating the mean opinion; can also be calculated from delay, jitter and packet-loss measurements to give an approximate score. A score greater than 4.0 equates to toll grade (see QoS).

Message Waiting
A feature that lights a message-waiting lamp to indicate receipt of a message (either voice- or text-based) from an attendant or another phone user.

Modulator/Demodulator (Modem)
A device that allows digital communication over circuits designed for voice by converting the data to suitable analog signals.

Multiple Trunk Groups
Allows central office trunk lines to be assigned to pools or groups in applications that require more than one central office trunk line for the same function.

Music-on-Hold
Audio signals (music, advertising or other sounds) that are played to users on hold so that they know they are still connected.

MySQL
An open source database that is able to run under various versions of UNIX, Windows and Mac systems and is often used to store and access provisioning system and subscriber feature data.

Network-Branch eXchange (NBX®)
A registered trademark of 3Com Corporation that identifies an IP telephony platform providing high-value converged services to small and medium-sized organizations.

Network Call Processor (NCP)
The heart of an IP telephony system that takes signaling from end points, such as phones and gateways, and forwards it to other endpoints based on dialing plans, routing rules and feature settings; configured by an administrator to provide the required IP telephony features.

Night Service
Allows central office trunk lines to be assigned in a fixed or flexible configuration to different ringing locations after normal business hours.

O—R

Off-Hook
The state of a telephone that is ready to receive dialing digits, in the process of dialing or has a call in progress (in contrast with on-hook); though normally determined by whether the handset is off the cradle, with modern speakerphones a device may go "off-hook" automatically as the first digit is pressed or upon pressing a dial or send button after entering digits. The term stems from the days when a telephone handset was hung on a hook.

Off-Premise Extension
Often a single-line phone, an off-site device that provides a user with the same capabilities and features as on-site phones.

On-Hook
The state of a telephone that is not ready for dialing or is in the process of dialing and does not have a call in progress (in contrast with off-hook).

On-Hook Dialing
Allows station users to dial calls without lifting the handset.

Outgoing Call Restrictions
Administrator-defined limitations placed on users or extensions that prohibit numbers or number ranges; often used to block external or international calling.

Packet Switching
A communications method that breaks-up data into small packets, routes them through the network and then reassembles them at the receiving end.

Personal Ringing
Allows users to program their phones with personal ringing tones to distinguish one ringing phone from another; may also be combined with Distinctive Ringing.

Phantom Mailbox
A user profile that assigns a telephone number but without an associated telephone and automatically connects all calls to its mailbox within a voicemail system.

Post Office Protocol 3 (POP3)
Common protocol enabling email clients to access email servers over an IP network.

Power Failure Transfer Phones
Allows specified central office trunk lines to be connected directly to certain phones in the event of a system or commercial power failure.

Primary Rate ISDN (PRI)
Service provider connection that delivers up to 24 (T1) or 30 (E1) switched data channels to a site. Each channel supports voice or 64 Kbps of data.

Privacy
Protects an existing conversation from accidental or intentional interruption by another user.

Privacy Release (Non-Privacy)
Gives a phone user control of the privacy feature, enabling its release should the user want to allow another phone user with the same line appearance to join the conversation.

Private Automated Branch Exchange (PABX)
(See PBX)

Private Branch Exchange (PBX)
Also referenced as PABX, Private Automated Branch Exchange, an inhouse telephony switching network that allows phones to be used for internal calls or calls to phones outside the network via a group of shared PSTN trunks. PBXs can support analog or digital services and capabilities such as least cost routing for outside calls, call forwarding, conference calling and call accounting.

Private Lines
Also known as personal lines, enable assignment of a trunk line to a specific button available only on certain telephones or to a specified phone user.

Protocol
Rules controlling communication between devices (rather like grammar in a language) for the purposes of exchanging of information.

Proxy Server (SIP proxy)
An intermediate device that receives requests from a client and then initiates requests on the client's behalf; a SIP proxy follows the rules of the Session Initiation Protocol (SIP)—standard RFC3261—to relay SIP signaling onto other proxies or clients.

Public Switched Telephone Network (PSTN)
The name of the international network of circuit-switched, automated telephone exchanges governed by International Telecommunication Union standards.

Q.931
Signaling protocol for interfacing with service provider networks over a T1 or E1 connection.

Q.SIG
An open, standards-based protocol that provides signaling and feature interoperability between PBXs.

Quality of Service (QoS)
Used to describe the characteristics required by a network in order to handle real-time communications and data applications simultaneously. Voice and video need QoS characteristics such as low delay, low packet loss and minimal jitter which are tolerated by most data applications.

Real-Time Protocol (RTP)
An IP protocol for voice, video and other real-time IP traffic that provides timestamping and synchronization information in the packet header to ensure proper reassembly when it reaches its destination.

Recorded Messages
Allows the system to connect callers to a recorded announcement in conjunction with such features as Night Service, Station Hunting, Automatic Call Distribution and Auto Wakeup.

Remote Access
Allows specified users to dial into the system from an outside location and be permitted access to system features and outside lines.

R—T

ReSerVation Protocol (RSVP)
A communications protocol for notifying a router that bandwidth needs to be reserved for real-time audio and video traffic to enhance transmission quality.

Saved Number Redial
Allows a user to dial a number and save it so that it can be redialed at a later time by pressing a button or dialing an access code.

Scheduled Conference
A pre-scheduled, formal meeting conducted as a call between multiple parties based on a predetermined participant list and/or a scheduled start and end time.

Screen Pop
A window that automatically opens on a user's computer when a predefined telephone event occurs; for example, an incoming call could generate a screen pop that lists caller ID information.

Session Description Protocol (SDP)
An IETF protocol that defines a text-based message format for multimedia session requests between IP devices using standards such as Session Initiation Protocol (SIP).

Secure Real-Time Transport Protocol (SRTP)
A protocal that provides confidentiality, message authentication and replay protection and control of RTP traffic.

Security Alarm Interface
Allows an alarm to trigger a telephone system alert whenever the door is opened.

Session Initiation Protocol (SIP)
An IETF application layer, text-based protocol, based on HTTP and MIME, that is used to handle session establishment between end points; enables interactive sessions to be established using SDP to negotiate required media such as text, voice or video.

Signaling
Indicators that control transmission of data or operation requests.

SIP Instant Messaging and Presence Leveraging Extensions (SIMPLE)
IETF SIP enhancements that provide a standardized method of delivering presence information and for supporting instant messaging (IM).

Simple Mail Transfer Protocol (SMTP)
A standard that defines the message format and transfer agent (MTA) needed for storing and forwarding emails using a variety of encoding methods to enable plain text, executable program and multimedia attachments.

Simple Network Management Protocol (SNMP)
A standard for sending messages between a management system and managed devices; includes the ability to interrogate device status and configuration, change parameters and receive alarms. SNMP-compliant devices and agents store data about themselves in management information bases (MIBs) and return this data to the SNMP requesters.

SIP Phone
An IP phone that uses Session Initiation Protocol (SIP)-based software for handling calls.

Station Message Detail Recording (SMDR)
Recording of data relating to incoming and outgoing calls that enables phone activity reports to be produced detailing time, duration and number dialed (see CDR).

Station Queuing/Trunk Queuing with Callback
Allows phone users who encounter a busy line to put themselves in a queue and go on-hook until called back when the number they called becomes available (see camp-on busy).

T1
A standard network connection, predominantly in the U.S., that can handle 24 channels with each channel able to handle 64 Kbps of data or a voice call. The total circuit operates at 1.544 Mbps.

T.38
A standard for converting a fax transmission from a circuit network to a packet network and vice versa.

Telephony
Voice communications.

Telephony Application Programming Interface (TAPI)
Call control and call manipulation standards developed by Microsoft and Intel for connecting computers to telephone services.

Tenant Service
Allows more than one organization (tenant) to share the same telephony system; through programming, each tenant can be restricted to its own central office (CO) trunks, attendant consoles and extension links; incoming calls are directed to the appropriate tenant.

Toll Restriction
Prevents specified phones from long distance toll calling (see outgoing call restrictions).

Toll Restriction Override
Enables authorized users to temporarily override toll restriction assignments on a specific phone by using a authorization code or specific abbreviated dialing numbers.

Traffic Measurement
Enables the system to generate feature and utilization reports on such statistics as number of calls, duration of calls placed on different lines or the frequency that all lines are busy.

Transmission Control Protocol/Internet Protocol (TCP/IP)
The family of protocols that supports communications via the Internet and most corporate networks.

Transport Layer Security (TLS)
Enables signaling to be encrypted over the network.

Trunk
A communications channel between two intermediate switching points in a network; often used to

T—W

refer to telephone channels between major switching centers or connections from a private network into a public network.

Trunk-to-Trunk Connections
Also known as Unsupervised Conference, allows a user to establish a connection between two outside lines without needing to remain in the conversation.

Unified Messaging
Provides access from a single interface—either a PC or phone—to voice, fax and email exchanges.

User Datagram Protocol (UDP)
Connectionless, unreliable transport protocol used over IP where error recovery is not required or is handled at a higher layer. Often used to carry RTP.

Virtual Tie Lines
Enable private connections between two or more telephony systems (two business sites) over an IP network.

Voice Core eXchange (VCX™)
A trademark of 3Com Corporation that identifies a SIP-based IP telephony platform providing high-value converged services to distributed organizations of any size.

Voicemail/Messaging System (VM/MS)
A standalone or networked system for receiving, storing and forwarding recorded voice messages.

Voice over Internet Protocol (VoIP)
A technology that leverages IP data networks for transporting voice calls.

Voice Profile for Internet Mail (VPIM)
A standard that enables software-based, seamless messaging between multiple sites and multiple voice messaging systems.

Voice Synthesizer
An electronic or software-based system that can generate voice prompts or messages by simulating a human voice.

Volume Control (for Microphone)
Allows adjusting the outgoing volume of a phone's microphone on a hands-free call.

Volume Control (for Speaker/Ringer)
Allows adjusting the volume of the incoming call ringer and the phone's speaker.

Wi-Fi Hotspot
A location providing access to a wireless LAN or the Internet.

Wi-Fi Phone
A wireless handset that uses the 802.11 standards to allow mobile telephony using Wi-Fi hotspots.

Visit www.3com.com for more information about 3Com secure converged network solutions.

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