VoIP Overview

VoIP Protocol Overview

The voice over IP (VoIP) protocol suite is generically broken into two categories, control plane protocols and data plane protocols. The control plane portion of the VoIP protocol is the traffic required to connect and maintain the actual user traffic. It is also responsible for maintaining overall network operation (router to router communications). The data plane (voice) portion of the VoIP protocol is the actual traffic that needs to get from one end to another.

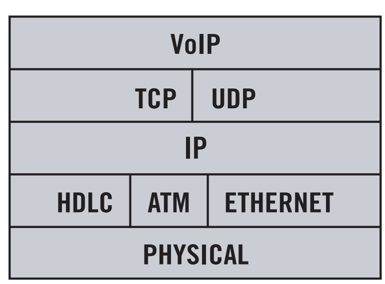
Common LAN traffic demonstrates a good example of differentiating between control plane and data plane traffic. A user may "surf the web" (HTTP) or send e-mail (SMTP) across a network. This constitutes data plane (user) traffic. On the other hand, the routers in that network are also communicating over the same LAN using OSPF (Open Shortest Path First) or RIP (Router Information Protocol). This traffic is never visible to the user, but it is required to route the user traffic. This constitutes control plane traffic.

Within the VoIP suite of protocols, voice packets are commonly referred to as the data plane. Likewise, signaling packets are commonly referred to as the control plane. This document will examine the VoIP protocol suite in this manner, data plane protocols and control plane protocols.

VoIP Protocol Stack

As its name implies, VoIP utilizes IP as its basic transport method. VoIP utilizes both the Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) over IP. The following diagram shows the protocol stack for a VoIP network.

It is important to note that VoIP works with any protocol stack that supports IP. End users of VoIP can add enterprise VoIP systems to their existing infrastructure relatively quickly and easily.



Data Plane Protocols (The Voice)

RTP and cRTP

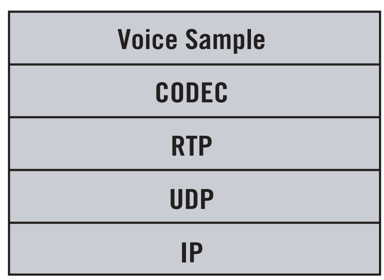
Both Real-Time Protocol (RTP) and Compressed Real-Time Protocol (cRTP) are currently available using any of the control plane protocols defined in this document. Since the voice traffic within a VoIP network can often take a different path than the signaling traffic, it makes sense that they are independent protocols.

RTP

RTP is the protocol that supports user voice. Each RTP packet contains a small sample of the voice conversation. The size of the packet and the size of the voice sample inside the packet will depend on the CODEC used.

RTP Protocol Stack

The following diagram shows the RTP stack.



RTP information is encapsulated in a UDP packet. If an RTP packet is lost or dropped by the network, it will not be retransmitted (as is standard for UDP). This is because a user would not want a long pause or delay in the conversation due to the network or the phones requesting lost packets. The network should be designed, though, so that few packets are lost in transmission.

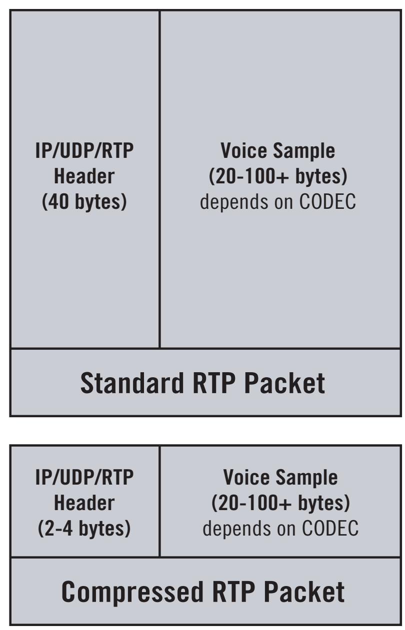
RTP Header

The RTP frame contains several pieces of information to identify and manage each individual call from endpoint to endpoint, which includes a timestamp, a sequence number, and conversation synchronization source information. A complete list of RTP header information is provided in Appendix A.

Compressed RTP

A variant of RTP is compressed RTP (cRTP), which eliminates much of the overall packet header. By eliminating this overhead, a more efficient packet is placed onto the network. With a system running cRTP, a user can place approximately twice as many calls as compared to a system running standard RTP.

Compressed RTP is used on point-to-point wide area network (WAN) links. Point-to-point, in this case, is not implying a PPP Layer 2 framing format. The link layer may be any standard WAN link layer protocol (frame relay, HDLC, PPP, or Cisco HDLC). The following diagram of a cRTP frame demonstrates why it can only be used on a point-to-point link.



Because the IP header is compressed with the UDP and RTP headers down to a maximum of 4 bytes, leaving no room for an IP address; therefore, the packet cannot be routed. It can only be placed on a point-to-point link that requires no addressing.

The issue with cRTP, similar to any form of compression, is that it takes more processing cycles of the router to process the packet. The router must recreate each header as the packet arrives so that the packet can be routed on the local area network (LAN) to the IP phone. When cRTP is used, router utilization can be a factor in the overall call quality.

RTCP

Real-Time Control Protocol (RTCP) is a data plane protocol that is not always used. This protocol allows the endpoints to communicate directly regarding the quality of the call. RTCP affords the endpoints the ability to adjust the call in real time to increase the quality of the call.

RTCP also aids significantly in the troubleshooting of a voice stream. Traditional VoIP analyzers sit at specific locations on a circuit and base their derived results from only the packets that they capture. With RTCP enabled, the analyzer can see the end-to-end quality as well as the quality at the point at which the analyzer is inserted, allowing the user to sectionalize problems much more quickly.

RTCP XR

RTP Control Protocol Extended Reports (RTCP XR) is a newer extension of the RTCP concept. It defines a set of metrics that can be inexpensively added to call managers, call gateways, and IP phones for call quality analysis. RTCP XR messages are exchanged periodically between IP phones and gateways.

With RTCP XR messages enabled, an analyzer sitting midstream on a voice call can see and decode the messages. These messages can also be retrieved via SNMP requests and can be fed into a larger network performance management system.

RTCP XR provides information on the following call quality metrics.

Packet Loss/Discard – The endpoints of a phone call examine each RTP packet and identify lost packets using the sequence numbers. The endpoints also identify those packets that arrive too late and are discarded by the endpoint. These RTP packets are referred to as discarded packets.

Delay – RTCP XR reports on the round-trip delay detected using RTCP and adds reporting information on the full envelope delay, which includes the CODEC and jitter buffer.

SNR and Echo – RTCP XR reports on the signal-to-noise ratio (SNR) at each endpoint. If the endpoint is equipped with an echo canceller, RTPC XR reports on the un-canceled echo level.

Overall Call Quality – Using simple embedded algorithms, RTCP XR can report MOS ratings or R-factor values for the call.

Configuration Information – RTCP XR can report on the overall configuration of an endpoint, including jitter buffer size.

CODECs

There is a wide range of voice CODECs (coder/decoder or compression/decompression) protocols available for VoIP phone implementation. The most common voice CODECs include G.711, G.723, G.726, G.728, and G.729. A brief description of each CODEC follows.

G.711 – Converts voice into a 64 kbps voice stream. This is the same CODEC used in traditional TDM T1 voice. It is considered the highest quality.

G.723.1 – Two different types of G.723.1 compression exist. One type uses a Code Excited Linear Prediction (CELP) compression algorithm and has a bit rate 5.3 kbps. The other type uses an Multi-Pulse - Maximum Likelihood Quantizer (MP-MLQ) algorithm and provides better quality sound. This type has a bit rate of 6.3 kbps.

G.726 – Allows for several different bit rates, including 40, 32, 24, and 16 kbps. It works well with packet to private branch exchange (PBX) interconnections. It is most commonly deployed at 32 kbps.

G.728 – Provides good voice quality and is specifically designed for low latency appli-cations. It compresses voice into a 16 kbps stream.

G.729 – One of the better voice quality CODECs. It converts voice into an 8 kbps stream. There are two versions of this CODEC, G.729 and G.729a. G.729a has a more simplified algorithm over G.729, allowing the end phones to have less processing power for the same level of quality.

Voice Quality Metrics

Overall Quality Factors

The following subsection describes several factors that can affect the quality of a VoIP call in an operational environment.

CODEC

The choice of CODEC is the first factor in determining the quality of a call. Generally, the higher the bit rate used for the CODEC, the better the voice quality. Higher bit rate CODECs, however, take up more space on the network and also allow for fewer total calls on the network.

Network

The biggest factor in call quality is the design, implementation, and use of the network that the voice calls are riding on. A typical VoIP call will start on a LAN at a CPE, go through a WAN connection to a provider cloud, and then go back out the other end. The CPE LAN and WAN links are most vulnerable to overuse and errors. Most VoIP quality issues are typically caused at these links.

There are several ways a network can affect a VoIP call, including packet jitter, packet loss, and delay.

Packet jitter – Jitter caused by changes in the inter-arrival gap between packets at the endpoint. The packets should arrive evenly spaced to allow for a seamless conversion into analog voice. If the packet gap changes, the user could experience degradation in quality. If the packet gap gets sufficiently large, the phone's packet jitter buffer will not be able to wait for the late packet, and the phone will drop the late packet. There are three different types of packet jitter – RFC jitter, instantaneous jitter, and absolute jitter.

RFC jitter, or "smoothed jitter," is defined by an ITU standard and essentially assigns a standardized value to the packet jitter of a call. The advantages of this metric are that it is defined by a standard organization and the equipment measuring this type of jitter should generate the same results. The disadvantages of RFC jitter are that it is a fluctuating average, and it eliminates spikes in the jitter that can cause packets to be dropped by the phone's jitter buffer. For these reasons, RFC jitter is not a very useful statistic.

Instantaneous jitter is the actual inter-packet jitter measurement, measuring the arrival time of each packet. There is no smoothing algorithm to eliminate spikes. Instantaneous jitter is the most realistic jitter measurement. The jitter buffer uses the instantaneous jitter measurement to determine which packets it will keep and which packets it will drop.

Absolute jitter is very different from RFC jitter or instantaneous jitter. Both RFC and instantaneous jitter rely on the current packet gap to determine their values. Absolute jitter represents the changes in inter-packet arrival times as compared to the previous packet gap.

Packet loss – The actual loss of voice packets through a network. Packet loss is often caused by congestion at one or more points along the path of the voice call or by a poor quality link (one that experiences physical layer errors).

Delay – Sometimes referred to as envelope delay, refers to the time it takes for the voice to travel from the handset of one phone to the ear piece of the other phone. Envelope delay is the sum of the delay caused by the CODEC of choice, jitter buffer in the phone, and the path time it takes for the packets to get through the network. A large delay can make conversation difficult.

Echo

Echo is a common problem for VoIP networks. It is important to note that, unlike packet jitter, packet loss, and delay, echo is not caused by the IP network. Echo is an analog impairment. It is extremely difficult to passively monitor for echo. The best way to detect echo is by placing a call onto the network with a known "good" device or capturing the voice packets of a call and playing them back for analysis.

There are two types of echo on analog voice networks – hybrid echo and acoustic echo. Hybrid echo is generated by impedance mismatches at various analog or digital points on the network. The most common location for the generation of hybrid echo is at a 2-wire to a 4-wire conversion point. Acoustic echo is generated at the phone. It occurs when the voice leaving the speaker is picked up by the microphone.

Measuring Quality

The following subsections describe the various methods for measuring voice quality on a VoIP network.

Intrusive: Non-real-time, two-ended methods

These methods involve sending known voice samples across a network from one endpoint to a receiving endpoint. The receiving endpoint does a comparison analysis of the degraded sample with the original. Because of the complexity of the signal comparisons, intrusive testing algorithms are computationally intensive and are not viable for real-time quality measurements. The following section describes the most common algorithms for this comparison.

PSQM

Perceptual Speech Quality Measurement (PSQM) is designed to avoid the subjective nature of Mean Opinion Score (MOS) rating and the effort it takes to get people into a room and listen to voice calls. PSQM measurements are performed by generating a known signal into the phone and then measuring what comes out the other end (post CODEC). The two signals are compared, and a PSQM value is derived.

However, PSQM was only designed to test the compression/decompression of CODEC functions. The algorithms that are used do not support overall end-to-end call quality through the network. Basically, PSQM cannot account for the effects of packet loss and packet jitter on voice quality.

PESQ

Perceptual Evaluation of Speech Quality (PESQ) was developed to expand the PSQM measurement from CODEC analysis to include distortion, filtering, and other channel impairments. PESQ is incapable of handling all of the issues that could occur in a network, including excessive delay and packet loss.

PAMS

Perceptual Analysis and Measurement System (PAMS) is similar to PSQM and PESQ, but it is designed to access the signal at an analog interface. Like PSQM and PESQ, a known signal is injected into the system, and the output is measured. It is not designed to test the overall end-to-end quality of a call.

Passive: Real-time, single-ended methods

These methods passively calculate voice quality without a reference voice sample. They are most commonly used in the turn-up and testing of actual networks.

E-Model (R-Factor)

The E-model produces a single value called an R-factor, which is derived from a variety of factors, including delay and other network impairments. Originally the E-Model was intended for use in network planning and design. The goal of the E-Model is to measure MOS without using all of the people that are typically required to provide an accurate MOS rating. R-factors range from 0 (extremely poor) to 100 (high quality). Any R-factor below 50 is unacceptable. TDM-based phone calls have a maximum R-factor of 94.

There are three main variations of R-factor – R_{CQE}, R_{LQE}, and R_{NPE}.

- R_{CQE}: The the call quality estimate is the estimated quality of the call in both directions.
- R_{LQE}: The listening quality estimate is a metric that removes delay impairments.
- R_{NPE}: The network performance estimate is a metric that removes CODEC degradation impairments, allowing the user to determine how the network is handling the raw packets.

Refer to Appendix E for a more detailed description of the calculation of R-factor, including the different variables in the equation.

MOS

MOS (Mean Opinion Score) assigns a value to the overall quality of the delivered voice through a network. This measurement scheme takes into account both the CODEC and the network. MOS ratings can range from 1 (bad) to 5 (excellent). A true MOS rating is determined by people listening to the same call and rating it from 1 to 5.

Test devices can measure MOS ratings through complicated algorithms based on the data from large groups of listeners rating calls. The test devices can then provide overall and per-call MOS ratings to give network operators an accurate view of how their network is performing. This is currently the most common VoIP call quality measurement.

The MOS score of a call can be increased using the packet loss control (PLC) algorithm, which can be applied at the phone or at the media gateway to mask packet loss to the end user. For low levels of packet loss, the algorithm detects the lost packets and plays back a small sample of speech, typically from the last received packet, which effectively tricks the end user's ear, masking the packet loss; thus affecting the overall MOS score.

Control Plane Protocols (The Signaling)

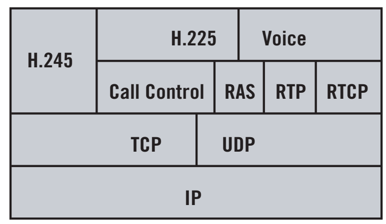
The control plane is used for the various signaling protocols, allowing users of VoIP to connect their phone calls. There are several different types of VoIP signaling available today, including H.323, SIP, SCCP, MGCP, MEGACO, and SIGTRAN. This document discusses the most prevalent types of signaling protocols today, H.323 and SIP.

H.323 Standard Overview

H.323 was the first widely adopted and deployed VoIP protocol suite. The H.323 standard was developed by the International Telecommunications Union-Technology Standardization Sector (ITU-T) for transmitting audio and video over the Internet. Over the past 10 years, this standard has gone through several revisions and additions to encompass more features, scalability, and stability. The current version of H.323 is Version 5.

H.323 Protocol

The overall protocol stack for H.323 (see below) is made up of several parts. Each part is responsible for specific tasks, such as call setup and phone registration.



H.245

H.245 is the media control portion of the H.323 protocol suite that establishes a logical channel for each call (endpoint to endpoint). During H.245 negotiation, each endpoint exchanges its capabilities and preferences. The choice of CODEC for the call is part of this exchange.

H.225

H.225 represents the basic signaling messages that are also used when dealing with ISDN or GR-303. For H.225, these messages include setup, alerting, connect, call proceeding, release complete, and facility messages that are based on the Q.931 signaling scheme as defined as below.

Setup – Message that attempts to connect a call. The calling party sends this message to the called party.

Alerting – Message that is sent from the called party back to the calling party to let the caller know that the far end is being alerted (ringing).

Connect – Message that informs the calling party that the called party has accepted the call. The conversation can begin at this point.

Call Proceeding – Message that informs the endpoints that the call is up and running. Call Proceeding messages are exchanged at specific intervals during the call.

Release Complete – Message that is sent by the party (called or calling) who disconnects the call first.

Facility – Messages that represent various control messages. They are often seen when a gateway is required to connect a call.

RAS

RAS (registration, admission, and status) protocol deals with element (phone) management. The RAS logical channel is established between the IP phones and the gatekeeper that manages those phones. Without appropriate RAS communications, an IP phone cannot place or receive calls.

Appendix B provides a more complete list of RAS and H.245 messages along with a sample of a basic signaling ladder.

SIP Overview

Session Initiation Protocol (SIP) is designed to manage and establish multimedia sessions, such as video conferencing, voice calls, and data sharing. SIP is still in its early stages of deployment and is a growing and evolving protocol standard. This is the standard that many element manufacturers are using to develop products.

There are several key features of SIP that make it so attractive:

1. URL addressing scheme – Allows for number portability that is physical location independent. Addressing can be a phone number, an IP address, or an e-mail address. The messages are very similar to those used by the Internet (HTTP).
2. Multimedia – SIP can have multiple media sessions during one call, which means that users can share a game, instant message (IM), and talk at the same time.
3. It is a "light" protocol and is easily scalable.

The two components that make up a SIP system include user agents and network servers.

User Agents – User agents represent the phone (user agent client) and the server (user agent server). The user agent client (UAC) initiates media calls. The user agent server (UAS) responds to those requests for setup on behalf of the UAC. The UAS is also responsible for finding the destination UAC or intermediate UAS.

Network Servers – Network servers include redirection, proxy, and registrar servers. Redirection servers do not process calls and only respond with information containing the appropriate address of the next server. Proxy servers contain features of both a client and a server. The proxy server can receive requests and response messages. It can also adjust the header information prior to forwarding the request to the next proxy server or back to the user client. The registration server registers new clients in the database and updates other databases.

SIP Protocol

As with HTTP, SIP messages can be broken into two major categories, including messages from clients to servers and messages from servers back to clients.

Message Headers

Each message has a message header that identifies the message type, calling party, and called party. There are four basic message types.

General Headers – Applies to request and response messages.

Entity Headers – Provides information about the message body type and the length.

Request Headers – Enables clients to include additional request information.

Response Headers – Enables the server to include additional response information.

Appendix C provides more information regarding message headers.

Request Messages/Methods

Request messages, or methods, are somewhat similar to the Q.931 messages used in ISDN. Request messages are initiated by a client to a server. SIP, a "light" protocol, has only a few request messages that it uses to connect calls. The following definitions apply to SIP request messages.

Invite – An invite message, as the name implies, is a request from a client to speak to another client. It contains the media type and other capabilities of the client.

Acknowledgment – The response message to an invite message. It represents the final message in the invite process and the beginning of the media exchange (voice).

Bye – The message sent by either client to end a call. The server is the first to receive the bye message followed by the opposite client.

Options – Allows the client to collect information on other clients and the servers.

Cancel – Cancels any message exchanges that are in progress but not yet completed.

Registration – Registers a client with a server and allows the client to use the services on the network.

Response Messages

In keeping with the "light" design of SIP and its Internet friendliness, SIP designers borrowed most of the response messages from HTTP.

There are two categories of response messages, provisional and final. Provisional messages are sent during a request/response process as details are worked out. Final messages, as the name implies, are the final response messages to a series of request/response messages.

There are five classes of response messages, including success, client error, server error, global failure, and informational. Each message class has several message types. Specific response messages are listed in Appendix C.

Other Signaling Protocols

SCTP, TALI, MGCP, and SCCP are other protocols that perform signaling functions on a VoIP network. It is important to note that multiple signaling protocols can exist in some portions of the network.

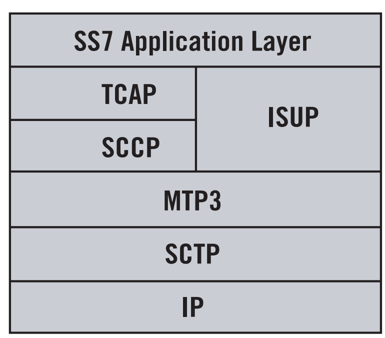
SCTP

Stream Control Transmission Protocol (SCTP) is a protocol format used for transmitting traditional TDM signaling protocols over IP. The main signaling protocol carried over SCTP is SS7 (this is also referred to as SIGTRAN).

Since SCTP carries traditional SS7 traffic, the protocol must meet the same guidelines defined for SS7. These guidelines include:

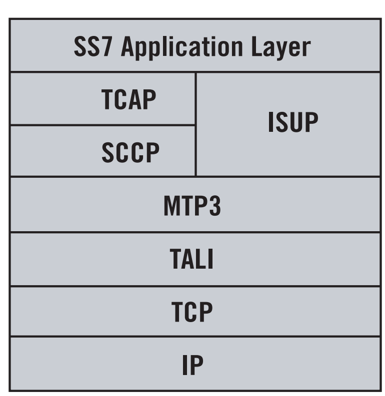
1. It must be compatible with UDP.
2. It must support acknowledged and error-free transfer of data.
3. It must support the segmentation of SS7 messages.
4. It must allow for network-level fault tolerance.

Since SCTP is essentially SS7 over IP, consult an SS7 guide to better understand this protocol and its message types. The SCTP protocol stack is as follows.



TALI

Transport Adaptation Layer Interface (TALI) is a standard very similar to SCTP. The TALI protocol is recommended for ISUP and TCAP messages transported over TCP/IP protocols. The TALI protocol stack is as follows.



As with SCTP, the 56 kbps DS0 with MTP1 and MTP2 layers are replaced by a new physical layer (typically Ethernet), MAC, IP, and TCP layers.

MGCP

Media Gateway Control Protocol (MGCP) is a combination of Cisco's Simple Gateway Control Protocol (SGCP) and IPDC (Level 3 protocol). The main feature of MGCP is the capability of breaking a telephony gateway into two basic parts, a call control and a media element.

In an MGCP system, there are a set of IP phones, a call controller, signaling gateways, and media gateways. The gatekeeper is a combination of a signaling gateway (SG) and a media gateway (MG). Signaling is converted from traditional PSTN signaling to a packet domain signaling protocol. Voice is converted from the PSTN G.711 CODEC to the CODEC of choice in the packet domain.

The CC manages the call routing. Multiple MGs are supported by one CC in an MGCP cloud, which allows for central management and control of the edge elements.

One of the main focuses of MGCP is the control and management of calls from one PSTN phone to another through the packet domain. Since the use of PSTN will not go away any time soon, the majority of calls over the next few years will be routed over an IP network to and from PSTN network elements.

SCCP

Skinny Client Control Protocol (SCCP) is Cisco's proprietary protocol used between a Cisco call manager and Cisco VoIP phones and is based heavily on the H.323 standard.

With SCCP architecture, the vast majority of H.323 processing power resides in the call manager. The end phones run the Cisco Skinny Client. The client requires very little processing power, which keeps the cost of the phones down and offers a more scaleable architecture than the standard H.323 architecture.

Testing VoIP Deployments

As carriers deploy VoIP services to their business customers, turning up and troubleshooting the service will become critical. Unlike traditional TDM services, a simple BER test will not fully qualify the service. The ability to place and receive calls for turn-up and monitor the calls for troubleshooting will be required.

Turn-up

To effectively turn-up a VoIP service, both signaling and call quality must be checked prior to customer hand off.

Signaling – A technician needs to place and receive calls through the network to make sure that the link is properly provisioned with the correct signaling protocol (H.323, SIP, etc). Calls should be placed within the VoIP cloud as well as from the VoIP cloud to the PSTN. These calls should include local and long distance calls to multiple exchanges.

By confirming that all possible types of calls can be placed, the technician can confidently connect the end-user's CPE equipment knowing that any signaling issues will not be within the carrier's cloud.

Call Quality – While checking the various call options, the technician can monitor the quality of the RTP stream. The two main values that the technician will want to examine are the R-factor (derived from the E-model) and an interpreted MOS rating. The technician can then compare these values with the SLA defined by the carrier.

Troubleshooting

Technicians must perform two types of troubleshooting: catastrophic failure and intermittent issues.

Catastrophic failure – When a circuit is non-operational, troubleshooting becomes very similar to turn-up. The technician can terminate the circuit back into a test device and begin the process of checking connectivity to the local elements through pings, trace routes, and phone call placement. The problem can be sectionalized to the CPE or carrier and then fixed by the appropriate party.

Intermittent issues – Intermittent issues represent the most difficult problems to solve. Since the error condition does not occur during every phone call, or even every day, the customer will most likely not allow the circuit to be taken down and tested. The technician must monitor the circuit and determine which calls have issues, when the issues have occurred, and what the environment is during those times.

It is absolutely **critical that the technician monitor every call on the circuit**. It is common for complaints to come in from a specific user representing one problem. The problem, however, may be from an element (CTMS or router, for example) or from a path in the network that affects many users on the network. An analyzer monitoring calls should be able to look at all of the calls at once so that larger problems can be identified. Depending on the CODEC and the voice sample time (ms) in each packet, a gigabit Ethernet link can process up to 27,000 calls simultaneously.

Voice Sample Time (ms)	Frame Rate (frame/s)	G.711		G.729	
		Voice Size (bytes)	Maximum Number of Calls	Voice Size (bytes)	Maximum Number of Calls
20	50	180	4,845	20	12,755
30	33.3	240	5,902	30	17,379
40	25	360	5,708	40	21,187
60	16.7	540	6,056	60	27,120

Environment, in this case, refers to the traffic riding on the network at the time of a failure. Other applications can use router processing time or even bandwidth, causing calls to drop or become difficult to listen to. The only way to determine if CPE traffic is the culprit of the VoIP issues is to monitor the whole circuit and measure call quality in real time.

When all of the factors (packet loss, jitter, and delay) for a VoIP call are added up, a call quality rating (either MOS-derived or R-factor) is calculated. Customers will base their quality complaints on a MOS rating using their ear. On the other hand, technicians will base their values on a derived R-factor rating using test devices. If the technician's derived R-factor rating and the customer's rating of the call differ, the call must be captured and listened to, which is the only way to successfully work with customers. Therefore, tools used for troubleshooting and turn-up must be able to play back voice and to emulate the actual VoIP phone being used.

Appendix A – RTP Header Information

The RTP Header has the following frame format.

	Bit 1	Bit 2	Bit 3	Bit 4	Bit 5	Bit 6	Bit 7	Bit 8
Byte 1	Version		Padding	Extension(X)	CSRC Count			
Byte 2	Marker			Payload Type				
Byte 3				Sequence Number				
Byte 4								
Byte 5								
Byte 6				Timestamp				
Byte 7								
Byte 8								
Byte 9								
Byte 10				SSRC				
Byte 11								
Byte 12								
Byte 13								
to				CSRC (0-60 bytes)				
Byte 72								

Version – The RTP version number, which is currently set to 2.

Padding – Gives the number of bytes at the end of the payload that are considered padding (not voice) and should be ignored. Padding is often used when encryption is enabled to keep the packets at a fixed length.

Extension (X) – If set, the header is extended.

CSRC Count – Provides the number of CSRC headers that follow the fixed header.

Marker – The interpretation of the marker is defined by a profile. It is intended to allow for the marking of significant events, such as frame boundaries, in the packet stream.

Payload Type – This field identifies the format of the RTP payload and determines its interpretation by the application. The following list contains the possible payload types, as defined by RFC3351.

Sequence Number – This number increments by one for each RTP data packet sent. In addition, it may be used by the receiver to detect packet loss and to restore packet sequence. The initial value of the sequence number is chosen randomly.

Timestamp – Reflects the sampling instant of the first octet in the RTP data packet. The sampling instant must be derived from a clock that increments monotonically and linearly in time to allow synchronization and jitter calculations. The resolution of the clock must be sufficient for the desired synchronization accuracy and for measuring packet arrival jitter. The clock frequency is dependent on the format of the data carried as payload. It is specified statically in the profile, or payload format specification, that defines the format. It may also be specified dynamically for payload formats defined through non-RTP means. If RTP packets are generated periodically, the nominal sampling instant, as determined from the sampling clock, is used. A reading of the system clock is not used. The initial value of the timestamp is random, as is the sequence number.

SSRC – Identifies the synchronization source. The value is chosen randomly, with the intent that no two synchronization sources within the same RTP session will have the same SSRC. Although the probability of multiple sources choosing the same identifier is low, all RTP implementations must be prepared to detect and resolve collisions. If a source changes its source transport address, it must also choose a new SSRC to avoid being interpreted as a looped source.

CSRC – A list identifying the contributing sources for the payload contained in the packet. The number of identifiers is given by the CC field. CSRC identifiers are inserted by mixers, using the SSRC identifiers of contributing sources.

Payload Type #	Name	Media (Audio/Video)
0	PCMU	Audio
1	Reserved	Audio
2	Reserved	Audio
3	GSM	Audio
4	G723	Audio
5	DVI4	Audio
6	DVI4	Audio
7	LPC	Audio
8	PCMA	Audio
9	G722	Audio
10	L16	Audio
11	L16	Audio
12	QCELP	Audio
13	CN	Audio
14	MPA	Audio
15	G728	Audio
16	DVI4	Audio
17	DVI4	Audio
18	G7209	Audio
19	Reserved	Audio
20	Unassigned	Audio
21	Unassigned	Audio
22	Unassigned	Audio
23	Unassigned	Audio
Dynamic	G726-40	Audio
Dynamic	G726-32	Audio
Dynamic	G726-24	Audio
Dynamic	G726-16	Audio
Dynamic	G729D	Audio
Dynamic	G729E	Audio
Dynamic	GSM-EFR	Audio
Dynamic	LS	Audio
Dynamic	RED	Audio
Dynamic	VDVI	Audio
24	Unassigned	Video
25	CelB	Video
26	JPEG	Video
27	Unassigned	Video
28	Nv	Video
29	Unassigned	Video
30	Unassigned	Video
31	H261	Video
32	MPV	Video
33	MP2T	Audio/Video
34	H263	Video
35-71	Unassigned	Not defined
72-76	Reserved	N/A
77-95	Unassigned	Not defined
96-127	Dynamic	Not defined
Dynamic	H263-1998	Video

Appendix B – H.323 RAS Messages, H.245 Messages, and Sample H.323 Signaling Ladder

RAS Messages

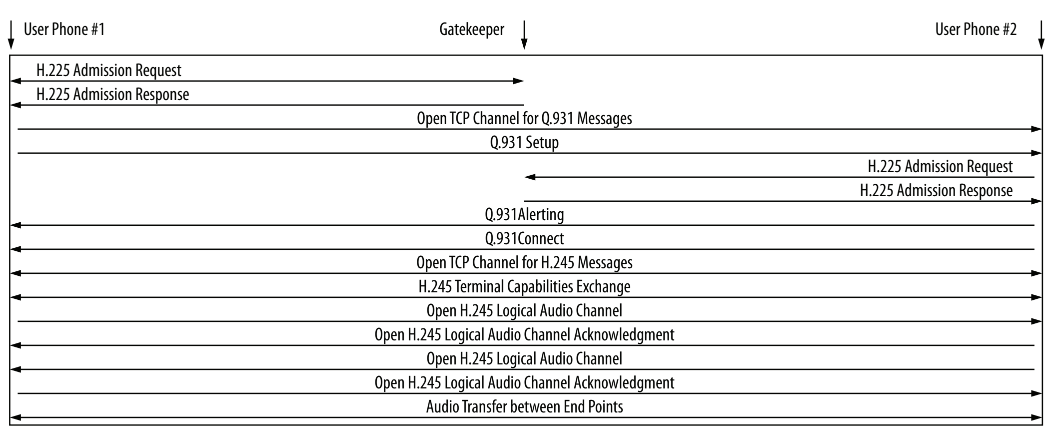
Message Type	*Message Name*	*Description*
RRQ	Registration Request	Sent from phone to gatekeeper to begin registration process.
RCF	Registration Confirm	Sent from gatekeeper to phone to confirm registration.
RRJ	Registration Rejection	Sent from gatekeeper to phone rejecting the registration request.
URQ	Unregister Request	Sent from phone to gatekeeper to unregister the phone.
UCF	Unregister Confirm	Sent from gatekeeper to phone to confirm unregistration.
URJ	Unregister Reject	Sent from gatekeeper to phone to indicate that the phone was never registered.
BRQ	Bandwidth Request	Sent by phone to gatekeeper to request an increase or decrease in the required bandwidth.
BCF	Bandwidth Confirm	Sent by gatekeeper to phone to confirm bandwidth increase or decrease.
BRJ	Bandwidth Reject	Sent by gatekeeper to phone rejecting the bandwidth change request.
LRQ	Location Request	Sent by phone to gatekeeper to request an endpoint address.
LCF	Location Confirm	Sent by gatekeeper to phone confirming an endpoint address.
LRJ	Location Reject	Sent by gatekeeper to the phone rejecting the endpoint address request.
ARQ	Admission Request	Sent by phone to gatekeeper to request call initiation.
ACF	Admission Confirm	Sent by gatekeeper to phone to confirm the call initiation request.
ARJ	Admission Reject	Sent by gatekeeper to phone rejecting the call initiation request.
IRQ	Information Request	Sent from gatekeeper to phone requesting a status update.
IRR	Information Response	Sent from phone to gatekeeper in response to a status request. Can also be sent periodically in the absence of an IRQ message.
SE	Status Enquiry	Can be sent from a gatekeeper to an endpoint or from one endpoint to another. Endpoints usually send these messages during calls for status. The gatekeeper can use these messages to keep track of active calls.

H.245 Messages

Message Type	*Description*
Capability Exchange	Consists of various messages that exchange the capabilities of the phones prior to a call being set up, including CODEC choice and media capabilities (video, data, or voice).
Master/Slave	For any call, one endpoint will be the master and the other will be the slave, which allows for the resolution of disputes when endpoints request features.
Round Trip Delay	Allows the phones to determine the delay across the cloud and back to the requesting endpoint. It also acts like a ping message. If the phone is not active, it will not respond to the round trip delay request.
Logical Channel Signaling	These messages open up the local channel for the RTP packets (voice) to travel across a network. If a gatekeeper is involved, the gatekeeper provides actual IP addresses of each endpoint so that the RTP stream can travel the shortest route.

Sample H.323 Signaling Ladder

This ladder assumes a single gatekeeper involved in the call. The ladder will be different if no gatekeeper is used or if multiple gatekeepers are used.



Appendix C – SIP Message Headers and Response Messages

SIP Message Headers

The following list contains the specific message headers in each SIP Message Header type.

SIP Message Header Type	Specific Header	
General Headers	Accept	Encryption
	Accept Encoding	Expires
	Accept Language	From
	Call ID	Record Route
	Contact	Timestamp
	CSeq	To
	Date	Via
Entity Headers	Content Encoding	
	Content Length	
	Content Type	
Request Headers	Authorization	Proxy Require
	Contact	Route
	Hide	Require
	Max Forwards	Response Key
	Organization	Subject
	Priority	User Agent
	Proxy Authorization	
Response Headers	Allow	Unsupported
	Proxy Authentication	Warning
	Retry After	WWW Authenticate
	Server	

Response Messages

Response messages, messages from the server to the client, are each assigned a unique status code that defines the message type. The status codes are grouped together by response classes.

Informational Responses

Status Code	*Message*
100	Trying
180	Ringing
181	Call Being Forwarded
182	Queued
183	Session Progress
200	OK

Success Responses

Status Code	*Message*
300	Multiple Choices
301	Moved Permanently
302	Moved Temporarily
303	See Other
305	Use Proxy
380	Alternative Service

Server Error Responses

Status Code	*Message*
500	Internal Server Error
501	Not Implemented
502	Bad Gateway
503	Service Unavailable
504	Gateway Time-out
505	Version Not Supported

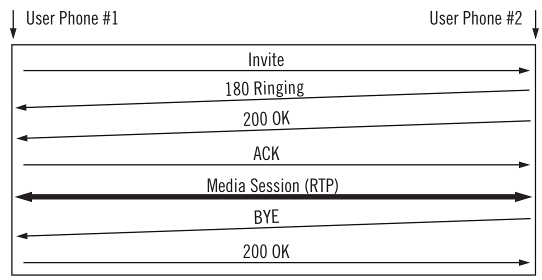
Global Failure Responses

Status Code	*Message*
300	Multiple Choices
301	Moved Permanently
302	Moved Temporarily
303	See Other
305	Use Proxy
380	Alternative Service

Client Error Responses

Status Code	*Message*
400	Bad Request
401	Unauthorized
402	Payment Required
403	Forbidden
404	Not Found
405	Method Not Allowed
406	Not Acceptable
407	Proxy Authentication Required
408	Request Timeout
409	Conflict
410	Gone
411	Length Required
413	Request Entity Too Large
414	Request URL Too Large
415	Unsupported Media Type
420	Bad Extension
480	Temporarily Not Available
481	Cell Leg or Transaction Does Not Exist
482	Loop Detected
483	Too Many Hops
484	Address Incomplete
485	Ambiguous
486	Busy Here
600	Busy Everywhere
603	Decline
604	Does Not Exist Anywhere
606	Not Acceptable

Appendix D – Sample SIP Signaling Ladder



Appendix E – R-Factor Determination

R-factor is determined using the following equation:

$$R = Ro - Is - Id - Ie\text{-}eff - Irecency + A$$

The components of the R-factor equation are described in the following table.

Component	Description
Ro	Signal-to-noise ratio.
Is	Combination of all impairments that occur simultaneously with the voice signal.
Id	Impairments caused by delay.
Ie-eff	Impairments caused by a low bit rate CODEC. Impairments due to packet loss and rejection.
Irecency	Impairments resulting from significant packet loss. Significant packet loss is detected if eight or more packets lost are in a row.
A	Advantage factor, which allows for the compensation of impairment factors.

Appendix F – Bibliography

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