Avaya FoIP, MoIP & TTYoIP

White paper & Application Note

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Fax and Modem over IP — FOIP/MoIP

Introduction

Despite the growth in broadband Internet Protocol (IP) network services and communication alternatives, the use of both facsimile (Fax) machines and modems remains strong. Today's businesses still depend on fax machines, fax servers, and modems that were designed to operate over Circuit-switched networks. By some estimates there are more than 100 million fax machines and 500 million modems worldwide. Standards that have enabled higher Internet data transmission speeds have helped to fuel modem use in computers, embedded devices, and standalone applications, while universal support of the Group 3 fax standard has taken the real-time advantages of fax communications and made it easy to use, robust over even very poor quality links, and widely available.

Our competitors in this market have various methods to address the use of these Time Division Multiplex (TDM) based products on an IP network. The introduction of a non-proprietary T.38 fax, proprietary Fax and Modem over IP (FoIP and MoIP) capability is keeping Avaya at parity with this market and the company leads the market with standards based communication accessibility by enabling section 508 compliance with TTY (Teletype) users with Text over IP (ToIP or TTYoIP).

The intent of these application notes is to provide an overview of fax, modem and TTY over an IP network and to address specific issues and concerns for implementing Fax, Modem and TTY within an Avaya platform.

Transmissions by fax, modem and TTY are mature technologies that were originally designed around Circuit-switched analog telephone lines. A paradigm shift in communication networking has moved these once well-behaved analog technologies into the world of data protocols. The shift is illustrated in the change in emphasis from:

- **Voice-centric** Circuit-switching [**Data-centric** Packet-switching
- **Dedicated and reliable** circuit path [**Variable and unpredictable** circuit path
- **High quality transmission** [Inherent **delay, jitter and packet loss**
- **Inefficient transmission** methods [**Increased transmission throughput**

The convergence of real time telephony applications into data switching and routing architectures presents many challenges, with the most important being to maintain the same user experience in terms of speed, quality and function while transmitting these signals over a data infrastructure. As with most advancement in technology, the customer's expectations have increased while the underlying technology has changed radically. It is no simple matter to transition these mature analog technologies into the data world and maintain the same customer experience. Fundamental system differences contribute to an increasingly complicated overall process.

This technical white paper will outline the key differences and underlying technical issues involved with bringing TTY, Fax, and Modem communication into the IP world.

Brief Networking Background

Circuit-switching vs. Packet-switching

The original world in which Fax, Modem and TTY machines were designed to operate was in the Public Switched Telephone Network (PSTN). The goal of this circuit-switched network was to enable a national telephony service that was both reliable and the highest quality. To achieve the high quality goal, a dedicated circuit was established before the first ring was heard at the other end. Once the circuit was complete or "nailed up," the voice conversation could use that static talk path. Since the entire route of the circuit was established across multiple switches and fixed prior to the call, no routing decisions are required during the communication and delay is very low. Although this provided high quality communication, the circuit-switching method was inefficient because both voice and silence were transmitted over the link (almost half of human speech is silence).

Packet-switching, on the other hand, is a data communications method originally designed to break up data files into packets of information for non-real time communication using no single predetermined path through a network. Any single "conversation" over a packet-switched network may use many separate links with packets guided through the network by routers. The routers use embedded digital source and destination addresses on each packet in the conversation, sending the information across links with shorter or longer delays based on link speeds and traffic encountered. Additional delay is introduced in packet-switching as routers buffer packets, encapsulate and reconstruct packets to read addresses, and provide error checking to ensure the original information is received accurately.

Overview of Fax, Modem and TTY

Since the Modem function is the common element in Fax, Modem and TTY machines it is important to understand key features of Modem technology. These machines were designed to use analog tones in a circuit-switched (analog) network to transmit digital information. TTY is used operationally in half duplex mode (only one party in a conversation can talk at a time), Modems are full duplex (both parties in a conversation can talk at the same time) and Fax begins and ends as full duplex while the transmission of the page is essentially half duplex. The following sections cover Modem and Fax issues together. TTY will be covered in a separate section.

What is a Modem?

A Modem (MODulator/DEModulator) modulates or "continually alters" a steady carrier signal to convert digital bits into analog tones. These analog tones are demodulated or interpreted by another Modem at the other end of the analog line to return the signal to its original digital format. Modems can use a variety of modulation techniques which are listed in Table 1.

Several other important elements of Modem technology are worth mentioning here:

- Modems are always used in pairs
- Digital information as a serial stream is the input to a Modem
- A Universal Asynchronous Receiver Transmitter (UART) within the Modem converts serial data into a parallel bus format
- A Codec (COder/DECoder) samples the analog signal and converts it into a digital signal — and vice versa
- Modulation is done by a Digital Signal Processor (DSP)

Modems also adhere to internationally agreed upon standards for encoding and transmission. Some of these standards are decided and published by the International Standards Union (ITU) in Geneva, Switzerland and are listed in Table 2.

Table 1 — Modulation Techniques	
AM	Amplitude Modulation
FM	Frequency Modulation
PM	Phase Modulation
FSK	Frequency Shift Keying
PSK	Phase Shift Keying
QAM	Quadrature Amplitude Modulation

Table 2 — ITU Encoding and Transmission Standards
V.22
V.25
V.32
V.34
V.90
V.92

Standards for Analog (Traditional) Fax

The ITU-T standard for analog Fax is T.30 and describes all the steps and timing sequences for Fax to work on the analog interface of the PSTN. A Fax machine is really two machines in one system — one sends and one receives. Both the sending and receiving functions consist of three complimentary components:

Sending Function	Receiving Function
Scanner — converts image to digital bits	Printer — print the transmitted image
DSP — compress the bits for transmission	DSP — uncompress bits to original image
Modem — convert bits to analog tones	Modem — convert tones to digital bits

A traditional Fax follows an established protocol for communication over an analog voice network. **Fax call setup** — the calling (sending) Fax machine emits a calling (CNG) tone of 1100 Hz for an *on-duration* of ½ second and an *off duration* of 3 seconds. This tone alerts people or other devices that a Fax call is initiating and should be directed to a Fax machine or other Fax receiving device.

The called (receiving) Fax machine emits a Called Station Identification (CED) tone at a steady 2100 Hz. If you mistakenly dial a Fax number, this tone instantly and annoyingly alerts you to that fact.

Fax transmission — the first step in transmission is where capabilities are negotiated such as compression format, bit rate, terminal identification, and synchronization. Then page transmission begins which is largely a half duplex operation. This makes sense because communication is primarily one-way, with pages transmitted from the sending machine to the receiving machine. After all pages are transmitted, more signaling takes place between the two machines to end the session and disconnect the call. One implicit, but important fact is that timing between signals is based on the low delay, jitter and loss guaranteed by the PSTN.

Types of Fax Equipment

There are seven internationally recognized specifications for Fax equipment (see Table 3). Of these, Group 3 Fax equipment (known as G3fe) is the most prevalent. A G3 Fax machine transmits at 9.6 Kbps. An enhanced version of G3 (EG3 or Enhanced G3) transmits at 14.4Kbps and has improved error correction.

The newest specification for analog Fax is called Super G3 (SG3) and has the capability of 33.6 Kbps transmission, V.8 handshaking, and T.85 image compression. This new standard results in speed improvement up to 6 times that of G3. Super G3 will negotiate down to standard G3 speeds and capabilities when connected to a G3 machine at the other end.

G4 is the beginning of an all digital Fax machine for use on a digital line — instead of analog. This specification eliminates the need for a modem which results in transmission speeds up to 64Kbps. These machines are still relatively rare in the US marketplace.

Table 3 — Types of Fax Equipment			
Type	**Year**	Speed	**Technology**
G1 — Group 1	1968	~6 minutes per page	Analog
G2 — Group 2	1976	~3 minutes per page	Analog
G3 — Group 3	1980	9.600 Kbps	Analog
EG3 — Enhanced G3	~1986	14.4 Kbps	Analog
SG3 — Super G3	1988	33.6 Kbps	Analog
G4	1988	64 Kbps	Digital

FoIP — Fax over Internet Protocol

Fax and modem traffic have evolved from transmitting tones over voice telephone lines, to digital signals that ride on data protocols such as TCP, UDP and IP. These signals are transported over both voice telephone lines and purely digital means of fiber, radio and other means through the Internet. However, the data protocols must still work within the analog timing structure.

FoIP adapts traditional Fax to use a new "carrier" as the IP data protocol over an enterprise LAN or even through the Internet. Simply put, FoIP enables analog Fax machines to interoperate with IP networks. Using FoIP, standard PSTN fax terminals can use an IP network as part of the end-to-end path and a Fax message can now be sent and/or received without an analog infrastructure and even without an analog Fax machine.

A Fax machine that can directly connect to an IP network is called an Internet Aware Fax (IAF) device. IAFs are a G4 type machine and use T.37 or T.38 protocols instead of analog T.30.

IP Standards T.37 and T.38

T.37 and T.38 are two common ITU standards that define using Fax over IP. There are others, such as T.434, T.6 and T.611 but they are not discussed in this paper.

T.37 specifies how a fax image is encapsulated in e-mail and transported to the receiver using store and forward technology. Most e-mail systems are store and forward systems where a mail server receives the e-mail and downloads it to your e-mail viewer at login. In this example, store and forward solves the problem of delivering email when the client PC is not online, without which the sender could only send email when the PC was turned on and online.

With T.37, Fax imaged documents are attached to e-mail headers and are encoded by the TIFF-F (Tagged Image File Format — Fax) compressed data format. In addition, there are two modes specified in the T.37 standard. Simple mode, specified in the S-profile, provides no confirmation of delivery. Full mode extension however, ensures call completion through the negotiation of capabilities and provides delivery confirmation. **Note that this standard was never intended to provide real-time service.**

Figure 1 — T.37 Store and forward system

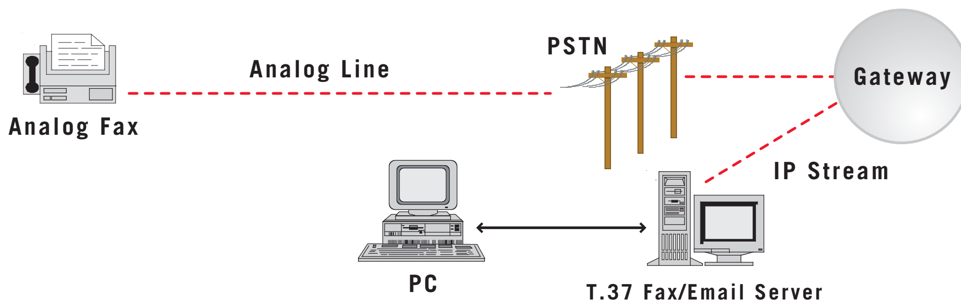
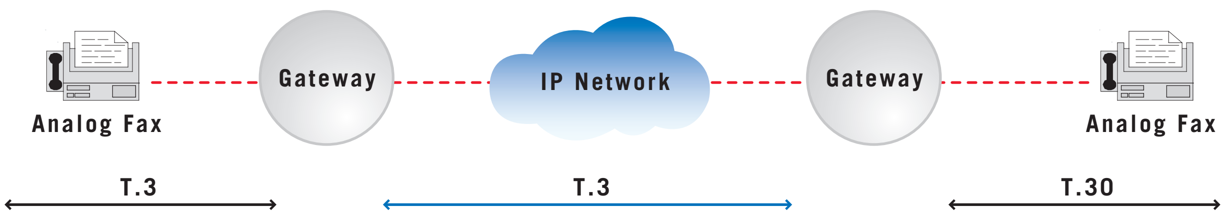


Figure 1 shows an analog Fax machine transmitting through the PSTN to a T.37 based server. This server stores the Fax and eventually sends the Fax message to the final destination (PC) after it is able to receive the message. This store and forward process is totally transparent to the sending Fax device.

T.38 describes a mechanism for transferring Fax in real-time over an IP link between standard Fax terminals (G3) or IAF devices. It also specifies a protocol by which IP Fax gateways or IAF devices exchange messages and data over an IP network. Gateways bridge the T.30 protocol (analog) to the T.38 IP protocol (digital). This is illustrated in Figure 2.

Figure 2 — T.38 Real-time (relay) system



Most FoIP systems use either a gateway or a native IAF device. Both devices commonly interface with the standard analog T.30 Fax protocol engine and couple it with a T.38 protocol module.

The T.38 protocol, like T.37, is also transparent to sending Fax devices. To achieve this, two methods are used — Fax relay and Fax Pass-Through.

Fax Relay

Fax Relay is one of two popular technologies to transfer a fax via the IP data protocol stack. Fax relay addresses the demodulation of standard analog Fax transmissions (T.30) from originating machines equipped with modems, transforming the analog stream into an IP data stream. This data stream is then "relayed" from the attached gateway using T.38 to the receiving gateway. The two gateways often have a data network and PSTN network between them. The receiving gateway converts the digital signal using T.30 standards to "re-modulate" to an analog signal for the destination Fax device. The diagram in Figure 2 is an example of a Fax relay system.

The process of transferring Fax documents by Fax Relay and IP gateways includes:

- Demodulating incoming T.30 signals from the sending Fax device
- Translating the T.30 fax signals into T.38 Internet Fax Protocol (IFP) packets
- Exchanging IFP packets between sending and receiving T.38 gateways
- Translating T.38 IFP packets back into T.30 signals at the receiving gateway
- Modulating T.30 signals and sending them to the receiving fax machine

Fax relay using T.38 must make the analog Fax machine believe it is talking to another fax machine. It does this by "spoofing" signals.

Fax Spoofing

Fax spoofing is another necessary mechanism to provide network transparency. The T.38 gateway transmits and receives Fax messages and control signals to trick the T.30 device into behaving as if it were directly connected (Circuit-switched) to another Fax device.

A T.38 (Packet-switched) network has undesirable characteristics that can interfere with the time sensitive T.30 originating device. This is namely delay, jitter, packet loss, and mis-order (packets arriving out of order). Spoofing can ameliorate the effects of delay, jitter and packet loss/mis-order by requesting padding image lines, or retransmitting a message to compensate for these network anomalies. (Note that these anomalies are virtually nonexistent in the PSTN because of the real-time nature of circuit-switched networks)

Fax spoofing is only partially effective because a Fax transmission can't be delayed indefinitely. Eventually absolute timers expire or critical signaling messages don't get through. In these cases the Fax transmission is aborted.

Fax Pass-Through

Fax Pass-Through, also known as voice band data mode, is the second popular method of transferring a Fax using the IP data protocol stack. This method uses a voice gateway or router to **pass** the fax machine's modem tones **through** the data and PSTN network to the receiving gateway or router and the eventual analog Fax machine. The Codec most used for this is a G.711 because it handles tones better than other compression Codecs. A Codec samples analog signals and converts them into digital signals. The G.711 Codec used for Fax Pass-Through is a special Codec that disables echo cancellation and Voice Activity Detection (VAD). This Codec is sometimes called a Fax Codec or a Clear Channel Codec.

The process of transferring Fax documents by Fax Pass-Through and IP gateways includes:

- Receiving the incoming T.30 signals from the sending Fax device
- Setting up a "clear channel" Codec
- Passing the tones through the pipeline of the data network/PSTN between sending and receiving gateways
- Sending the original tones to the final destination Fax machine

Data Network Attributes

User Datagram Protocol vs. Transmission Control Protocol

FoIP can be carried over two common transport layer protocols — UDP and TCP.

UDP, or User Datagram Protocol is a connectionless method that transmits IP packets to the destination device. UDP is a low overhead protocol that is ideally suited to real-time applications, but it will not retransmit lost or corrupted packets. In other words, it is fast and efficient, but does not have error correction capabilities to ensure packets arrive uncorrupted and in order. To compensate for this, some vendors send all FoIP messages two or more times, in case one of them is lost. This is called Redundancy. Redundancy significantly increases bandwidth to better ensure against lost packets. Avaya uses UDP for FoIP messages — Relay, T.38 and Pass-Through modes.

TCP, or Transmission Control Protocol is a connection-oriented protocol that can also transmit packets to the destination device. This protocol specifies error checking and retransmission of failed packets as needed. Since different links within IP networks have different data transmission rates, packets arriving out of order are stored in memory for reconstructing the exact packet stream at the receiving device. This quality of service takes time which makes this protocol nicely suited for T.37 Faxing or other non-real time services.

Delay

There is some delay in the Fax process because the scanner, DSP and modem all take time to perform their functions. FoIP introduces *additional* delay called packet delay. Packet delay is the length of time it takes a packet to traverse the LAN/WAN data network. Each element of the network adds to packet delay including switches, routers, distance traveled through the network, firewalls, and jitter buffers. Delay can interfere with T.30 signaling, which is why spoofing is so important.

Jitter

Jitter is a measure of variance in the time it takes for communications to traverse from the sending application to the receiving application. Jitter is a new anomaly introduced with FoIP because it travels over a data network. Data applications are almost always bursty, hence intermittent congestion in data networks is a main source of jitter. Too much jitter can, but may not, cause loss of packets if UDP transport is used and packet queues are sufficiently short.

Packet Loss

Network packet loss occurs when packets are sent, but not received at the final destination due to some network problem. Evenly distributed packet loss may only affect clarity of the Fax transmission, but the loss of several contiguous packets can terminate the Fax or Modem session because control signals may be lost and T.30 has timing limitations. Bursty packet loss is a more difficult environment because the chances are greater that several critical packets will be lost.

Mis-Ordered Packets

Conditions exist in IP networks for packets to take different paths, and therefore, those packets can arrive at the destination or gateway out of order. Mis-ordered packets can be dropped when queues are full and at best, the fax quality suffers. At worst, signaling can be lost and the Fax message or Modem session can terminate prematurely. Techniques like load balancing over unequal links can cause mis-ordered packets.

Packet number sequencing in fax/voice provide for the ability to reorder packets, assuming delay isn't too great and jitter buffers don't deplete.

FoIP Recommendations

Transcoding compresses and decompresses the Fax stream by changing it from analog to digital through a codec. FoIP must do this between gateways to enable the stream to travel over an IP infrastructure. It may be difficult or impossible to use FoIP if more than one set of gateways are in the total end-to-end path of the Fax transmission because of multiple transcodings. More transcodings in a path cause more delay and the T.30 timers may not wait for that delay. Therefore only one set of gateways is recommended for both relay and pass-through schemes.

Fax Packet Loss should be less than 1% if evenly distributed and no more than 2 consecutive packets if bursty, with no redundant packets. More loss can, but may not cause the Fax call to terminate. Losing picture transmit packets degrades fax quality, but losing just two signal packets can result in an aborted fax message. Network measured packet loss can approach 7% before quality degrades and 10% before Fax fails — using redundant packets, but again, only if consecutive control (signal) messages are not lost.

Modem Packet loss is only slightly more forgiving.

Delay one-way, through the end-to-end network should be no more than 500 milliseconds because of T.30 timing requirements for both FoIP and MoIP. This limit includes more than just the delay in the network — it includes all components from end-to-end.

Jitter through the network should be not equal to or greater than the codec payload size. If the codec uses 20ms packets, jitter should not be equal or more than 20ms.

Acceptable Fax machine types include G3, Enhanced G3 and Super G3. No others are supported including G4 Internet Aware Fax (IAF).

Quality of Service (QoS)

It is highly recommended to apply QoS to any real-time application. Fax, like voice, is considered real-time for both Fax Relay and Fax Pass-Through because of T.30 timing limitations. QoS means that the network recognizes specific data packets as important and the network should therefore move these packets ahead of others. If congestion is present, data packets should be discarded before any voice or fax packets are discarded. Data packets can be resent since they are not real-time traffic and are bursty in nature, using protocols such as TCP to handle error correction and retransmission to ensure transmission accuracy.

Bandwidth Requirements

The WAN bandwidth needed for FoIP depends on the method used — Fax Relay or Fax Pass-Through

- **Fax Relay** bandwidth is based on the negotiated Fax modem speed and is around 27K per second. Avaya Relay and T.38 uses 30ms packets. The relay bandwidth formula is (payload + RTP + UDP + IP + Layer-2 overhead) *8 *33.

  36B Payload + 12B RTP + 8B UDP + 20B IP + 18B MAC + 8B Preamble = 102 Bytes per second.

  102 Bytes * 8 (convert to bits) * 33.333 (packets per second) = **27.2K** (for each direction)

  For single redundancy, double this value.
  For double redundancy, triple this value.
  For triple redundancy, quadruple this value.

- **Fax Pass-Through/Modem Pass-thru/TTY pass-thru** bandwidth is based on the header overhead of T.38 + UDP + IP + L2. The T.38 payload is dependent upon the “voice” payload defined for Voice over IP (VoIP). The default is 20 milliseconds. An example for default settings using an IP trunk between gateways would be [160B payload + 12B RTP + 8B UDP + 20B IP + 22B L2 + Preamble] = 230 Bytes.

  230 * 8 * 50 = **92Kbps** (per each direction). This is for 20ms packets. 1, 2 or 3 redundant packets can double, triple or quadruple this one-way bandwidth.

Security

Both Fax relay and Fax Pass-Through can become secure by using techniques that are available with today’s technology.

The relay (or T.38) portion of Fax Relay is essentially a TCP/IP protocol stack riding on a data layer-2 protocol like Ethernet, PPP, ATM, etc. Like any data stream, this T.38 Fax stream will be encrypted if the data link uses a VPN (Virtual Private Network). VPNs usually encrypt using 3-DES but can also use other means of encryption.

- Pass-Through can also be encrypted independently of VPN using either Avaya's proprietary AEA encryption or the industry standard AES
- Avaya Relay can be encrypted using Avaya's AEA method
- T.38 cannot be encrypted by either AEA or AES

Legal Implications of leaving traditional Fax

Fax is well used and sanctioned by the U.S. and International legal systems. Newer technologies, such as flatbed scanners and email can transport the same documents with better quality, probably less time and perhaps better security if using VPNs. Even though superior technologies are available, the U.S. legal system is slow to adopt them. This prolongs the life analog Fax.

Future viability of Fax over anything

Fax is a mature technology to transmit type and images of the printed page over analog PSTN lines to a destination point. FoIP is more complicated and introduces more limitations to force this older method to work on newer digital networks. Scanners and email are far more efficient and may eventually replace Fax over any architecture. However, Fax capability is very important in business and especially in countries that do not have a large base of PCs in use. Therefore, Fax will be with us for a long time and the need to continue to embrace and enhance it will undoubtedly remain.

Text over IP — ToIP

Text over IP refers to TTY (or TDD) that is the transmission of the typed characters from a TTY device over an IP network. TTY machines are small keypads that connect directly to an analog telephone line. TTY devices help enterprises comply with equal-access regulations (i.e. Americans with Disabilities Act, Workforce Investment Act) by enabling them to meet the needs of their hearing/speech-impaired employees.

TTY Standards

There are national and international standards for TTY. TTY uses "Baudot Tones". A Baud is a change between two or more frequencies. T.140 describes the mechanisms for providing reliable real-time text conversations over IP systems. The most commonly used TTY standard in the US is the ANSI/TIA/EIA 825 ("A 45.45 Baud FSK Modem"). The UK uses a 50 Baud standard.

Differences of ToIP from FoIP and MoIP

There are three key differences between ToIP and FoIP or MoIP. These differences are explained here:

1 **TTYs are silent when not transmitting.** — Unlike fax machines and computer modems, TTYs have no "handshake" procedure at the start of a call, nor do they have a carrier tone during the call. Although this approach tends to limit the speed of transmission, it has the advantage of permitting TTY tones, DTMF tones (Dual Tone Multi-Frequency, *aka* touch tones), and voice to be intermixed on the same call.

2 **TTY operation is half duplex.** — TTY users must take turns transmitting, and typically cannot interrupt each other. If both people try to type at the same time, their TTYs will show no text at all, or will show text that is gibberish. There is no automatic mechanism that lets TTY users know when a character they have typed correctly has been received incorrectly.

3 **Each TTY character consists of a sequence of seven individual tones.** — The first tone is always a "start tone" at 1800 Hz. This is followed by a series of five tones, at either 1400 or 1800 Hz, which specify the character. This yields 32 possible characters for each of two modes or 64 total characters. All letters, numbers and special characters are available.

- The final tone in the sequence is always a "stop tone" at 1400 Hz. The "stop tone" is a border that separates this character from the next.
- Each of the first six tones is 22 milliseconds in duration (20 milliseconds for 50 Baud). The final "stop tone" may also be 22 milliseconds, but is permitted to be as long as 44 milliseconds. This means that the duration of each TTY character is at least 154 milliseconds, which work out to approximately six and a half characters per second. (The description of this as a "45.45 Baud" protocol is based on the number of 22-millisecond tones that can be transmitted in one second, not the number of characters.)

Transmission Problems

Many of the techniques that are commonly employed by telephone systems to digitize audio signals (such as voice) are able to digitize TTY tones with perfect accuracy. Unfortunately, some techniques that are optimized for low-bit-rate encoding of voice signals tend to distort TTY tones such as GSM encoding for cellular phones.

The discussion of VoIP below assumes that a TTY-compatible technique, such as G.711 or G.726, is being employed. The essential point of the discussion is that it's the transmission techniques, and not the audio encoding techniques, that are the primary barrier to TTY compatibility in VoIP systems.

VoIP systems transmit audio streams, such as voice and TTY tones, by digitally encoding the audio and then breaking the streams into individual packets. (A typical packet contains a 20-millisecond stream of audio. Each of these audio packets is tagged with header information including a sequence number and the destination IP address. The complete packet is then delivered by the originating device to the network, which transports the packet via shared pathways that often contain packets from many different sources, with many different destinations.

Although the destination is specified in the packet header information, the route to the destination is not specified. The ability for each packet to take what is, at that instance, the "best" route to its destination is where VoIP derives a lot of its economic advantage. It's also the reason why TTY-on-VoIP can be unreliable: because packets are free to take different pathways, they cannot be relied upon to arrive at the receiving device before it's their "turn" to be played. Although these packets often arrive eventually, they are regarded as lost because they did not arrive in time, and must therefore be discarded.

Under most circumstances, the loss of occasional packets is not detectable in voice communication. VoIP telephones employ packet loss concealment algorithms that trick the human ear, typically by mimicking the adjacent packets that have been received. Although these techniques work well with voice, they do not work with TTY tones. If a packet containing a TTY tone is lost, the current generation of VoIP techniques is unable to recover it or rebuild it.

Packet Loss

What is the impact of packet loss on TTY? Consider the following illustrative example:

> Assume that the VoIP packet size is 20 milliseconds (a typical value) and that the packet loss rate is 0.5% (a rate generally regarded as excellent for VoIP communication). Keep in mind that an individual TTY text character is at least 154 milliseconds in length, and therefore spans eight packets. This means that, if there is a 0.5% likelihood that any one of those packets is missing, approximately four percent of all TTY characters will lose one of their packets. If any one of the eight packets within a character is lost, that character will not be displayed properly on the receiving device.

Even though the simple statistical model above would seem to predict a four percent TTY error rate under the described conditions (20-millisecond packet size, 0.5% packet loss rate), the actual error rate would tend to be much higher. This is because, if the lost packet is the one that contained the "stop tone" for that character, subsequent characters, **even if transmitted without packet loss,** might nevertheless be decoded improperly.

As a point of comparison, a TTY character error rate of more than one percent is generally regarded as unacceptable, chiefly because the transmission of information such as bank balances and credit card numbers becomes unreliable.www.atis.org/pub/ttyforum/tty19-01-09-26-11.pdf explains TTY user requirements. Using a simple statistical model that is based on a 20-millisecond packet size, and ignoring the additional deleterious effects that result from dropping a "stop tone," the one percent character error rate threshold is exceeded when VoIP packet loss rates exceed approximately 0.12%.

Delay and Jitter

Delay and jitter are almost non-issues because even in a perfect world where no packet loss is seen, most users of TTY cannot type faster than the ceiling of 6 characters per second. This equates to 360 characters per minute. Keep in mind that the users act in half duplex. Typing at the same time the other TTY user is typing results in garbled messages. Therefore, operationally ToIP is slower than MoIP or FOIP and slow to the point that even delays of 1 or 2 seconds can be acceptable.

Appendix A

Traditional (analog) Fax using the T.30 Protocol

The International Telecommunications Union (ITU) standard that defines Fax is T.30.

T.30 defines the phases necessary for document transmission between two fax terminals in the Public Switched Telephone Network (PSTN). There are five phases:

Phase A Call Establishment	**Phase B** Pre-message Transmission Function	**Phase C1** In-message Function **Phase C2** Message Transmission	**Phase D** Post-message Function	**Phase E** Call Release

← Facsimile Function →

← Facsimile Call Duration →

Phase A — A call is established after a connection exists between the sending and receiving terminal and Fax tones can be exchanged. Calls can be manually or automatically setup. In manual operation, a person dials the Fax telephone number and hears the familiar telephone ring called ring-back. After the called endpoint answers, the person hears the Fax calling tone. The person then switches from their handset to the fax endpoint to continue the transmission. In automatic operation, the Fax endpoint detects dial tone, and dials the desired telephone number. When the called Fax endpoint answers, the calling endpoint immediately transmits a Fax calling tone.

Phase B — The pre-message function has both identification and command sections.

Identification involves:

- Capabilities of the called endpoint such as bit rate, page length, compression format, etc.
- Reception confirmation sent by the called endpoint.
- Terminal identification such as telephone number and name of organization, etc.
- Non-standard facilities identification such as enhanced security.

Command involves:

- Specifying chosen or negotiated capabilities that were identified.
- Training signal that tests the acceptability of the chosen data rate.
- Synchronization of the two Fax endpoints.

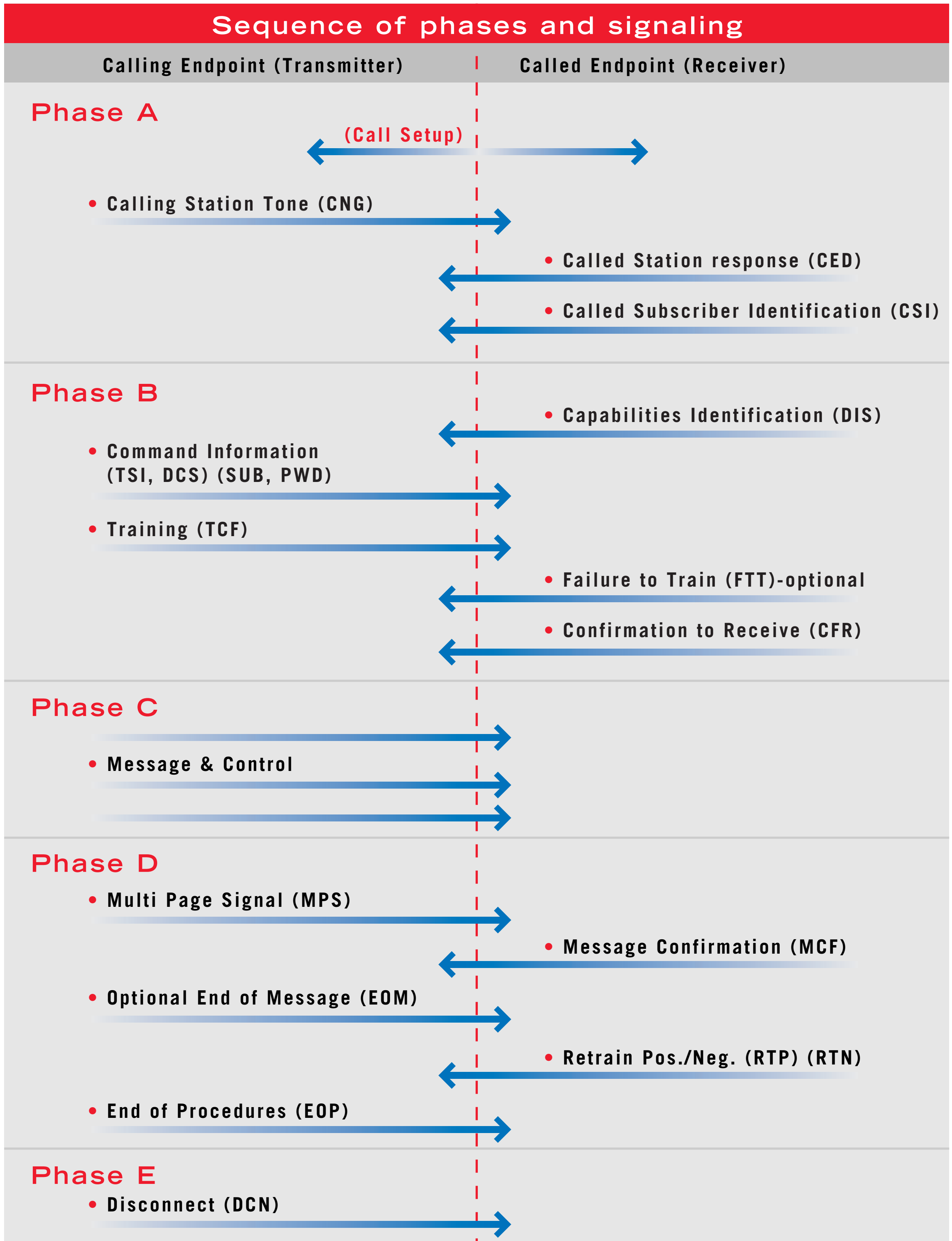
Phase C1 — The in-message function is the control signals that are sent along with the message. This includes error detection, correction, in-message synchronization and line supervision.

Phase C2 — The message transmission is the coding scheme of the message itself and is further defined by the ITU-T T.4 standard and is outside the scope of this paper.

Phase D — The post-message function includes:

- End of message signaling. All pages have been transmitted and a return to Phase B is possible if needed.
- Multi-page signaling. Sent by the calling endpoint to indicate a page of the document has completely transmitted and returns to Phase C2 to receive the next page of the document.
- End of procedure signaling. Sent by the called endpoint to indicate that all pages of the document have transmitted and no more are pending. The session can now proceed to Phase E.
- Confirmation signaling. Sent by the called endpoint to confirm any of the above messages.

Phase E — The calling endpoint terminates the connection by transmitting a special disconnect signal to the called endpoint. No response is needed or expected.


Sequence of phases and signaling
Calling Endpoint (Transmitter)
Called Endpoint (Receiver)
Phase A
(Call Setup)
• Calling Station Tone (CNG)
• Called Station response (CED)
• Called Subscriber Identification (CSI)
Phase B
• Capabilities Identification (DIS)
• Command Information
(TSI, DCS) (SUB, PWD)
• Training (TCF)
• Failure to Train (FTT)-optional
• Confirmation to Receive (CFR)
Phase C
• Message & Control
Phase D
• Multi Page Signal (MPS)
• Message Confirmation (MCF)
• Optional End of Message (EOM)
• Retrain Pos./Neg. (RTP) (RTN)
• End of Procedures (EOP)
Phase E
• Disconnect (DCN)

Appendix B

Application Note: Fax & Modem over Internet Protocol (FoIP/MoIP)

Support of Fax and Modem over Internet Protocol

Avaya Communication Manager enables a natural migration path from today's separate data and voice networks to a converged network environment that includes support for existing fax and modem-based services.

Avaya's Features, Speeds and Releases

Appendix B — Table 1

Technology	Feature	Maximum Rate	Release Supported
Fax	Avaya Fax Relay	9.6 Kbps	DEFINITY Release 10 and later
	T.38 Standards-Based Fax Relay	9.6 Kbps	ACM Release 2.1.1 and later
	Avaya Fax Pass-Through	33.6 Kbps	ACM Release 2.1.0 and later
Modem	Avaya Modem Relay	9.6 Kbps	ACM Release 2.2 and later
	Avaya Modem Pass-Through	33.6 Kbps or less likely, possibly up to 56 Kbps if over digital trunks and circuits.	ACM Release 2.1.1 and later
TTY	Avaya TTY Relay (US)	45.45 Baudot	ACM Release 2.0
	Avaya TTY Relay (UK)	50 Baudot	ACM Release 2.1.0
	Avaya TTY Pass-Through	Typically 45.45/50 Baudot but 64kbps channel is provided	ACM Release 2.1.1 and later

Fax Relay Types

The relay function supported in MV 1.3 is proprietary because the T.38 standard was not completed before Avaya offered it. *Beginning with ACM 2.1.1, there are now two "relay" functions — Avaya proprietary and the T.38 standard.* A fax relay is basically a tunnel between two gateways that the Fax or Modem signals travel across. The default relay method is the Avaya proprietary scheme. Pass-through essentially passes through an IP system like any other IP stream, but is dependent on low latency, jitter and packet loss.

Fax/Modem Pass-Through vs. Relay

There are several challenges to address with supporting Fax/Modem communications over IP. The original digital signal was converted to analog by the modem. This analog signal now must converted back to a digital format over IP. This requires sampling, which causes some degradation. If voice compression Codecs are being used (i.e. G729 or G.723) they can cause problems by distorting the fax/modem tones. The use of G.711 or G.726 helps to minimize this by providing greater sampling rate bandwidth.

Fax/modem transmissions are susceptible latency, jitter, and packet loss. Limiting the number of PSTN-to-IP and IP-to-IP hops in the path of the call reduces latency, jitter, and signal degradation issues, while the implementation of duplicate packets can help mitigate packet loss.

Table 2 compares Fax/Modem relay and Pass-Through methods

Appendix B — Table 2

Pass-Through	Relay
Gateways use a G.711 or future G.726 codec with no Echo Canceller (EC) or Voice Activity Detector (VAD) and static jitter buffers	Gateways involved in modulating/demodulating the fax/modem information talking both T.30 to the Fax machine and T.38 to the other gateway
Gateway passes modulated fax/modem information in-band over IP network as a voice call using a G.711	Near-End gateways demodulate fax/modem information into IP packets Far-End gateways remodulate IP packets into analog fax/modem signal
Uses more bandwidth, better suited for LAN	Uses less bandwidth, well-suited for WAN
Needs both ends of call to be synchronized to a common clock source	Works better than Pass-Through when there is no common clock source
Susceptible to packet loss	Provides packet redundancy protection options for packet loss
Requires less media processor resources	Requires more media processor resources

Note: Modem Relay is NOT supported until the release of ACM 2.2.

General Fax/Modem Call Flow for ACM 2.1.1 and later — (see Figure 1)

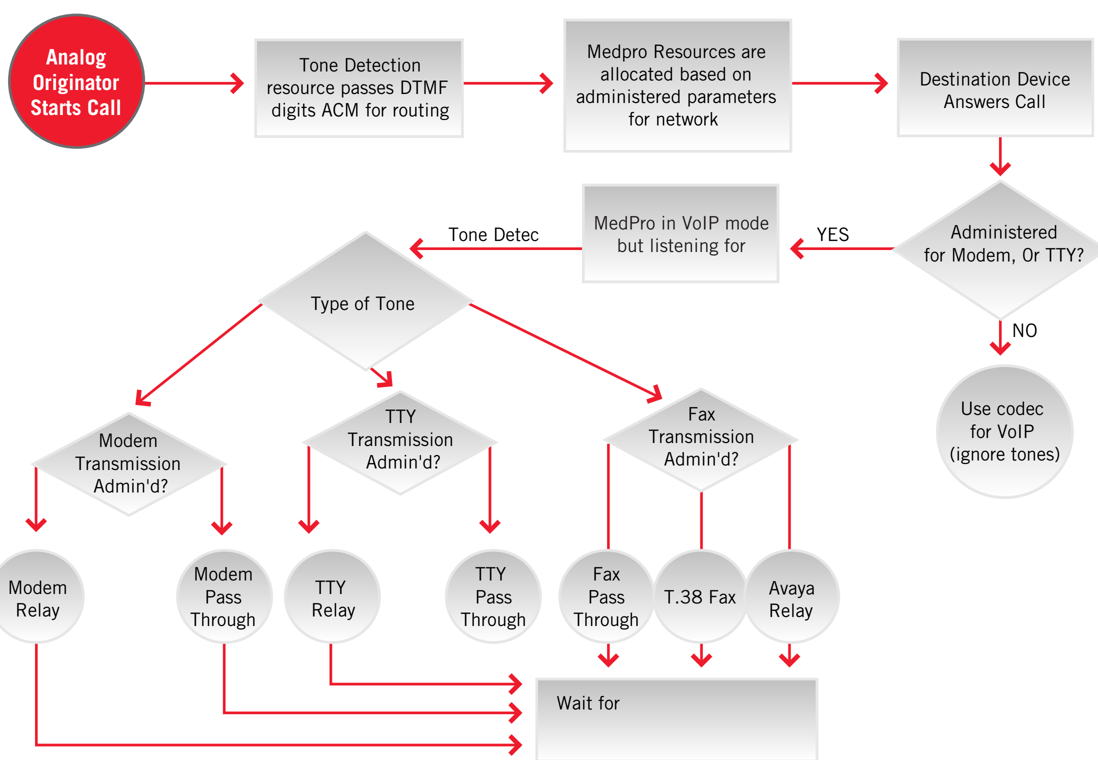
1. The originating analog device (Fax/Modem/TTY/STU) starts a call to a destination.
2. Communication Manager routes the call to the called number and establishes a call session as an audio call.
3. Once the audio call is established, fax/modem tones can be exchanged. These tones indicate whether the call is a Fax, Modem or TTY.
4. If the call is T.38 and is administered, the encoding of the call is changed to T.38 mode, and Communication Manager signals to the gateways to use T.38 procedures to negotiate a switch-over

to T.38 fax mode. If the gateway at the opposite end of the call acknowledges the T.38 mode request, the initial audio channel is closed and a T.38 Fax Relay channel is opened. When the fax transmission is completed, the call is reverted back to voice mode. Similar procedures are followed for Avaya proprietary fax/modem relay, which is the default mode between Avaya media gateways.

5. If the call is administered as Pass-Through, the media processor simulates a "clear channel" by switching the existing voice coder on the fly to G.711 data mode to increase bandwidth, removes VAD, echo cancellers and set buffers to reduce jitter.

6. At conclusion of call, resources are released for use by other calls.

Figure 1



Bandwidth and Resource Considerations

Generally, Relay is the preferred mode for supporting Fax/Modem over IP communications from a *bandwidth point of view* only. Relay requires significantly less bandwidth than Pass-Through (see table 3). It also compensates better for packet loss using packet redundancy, which sends redundant packets in addition to the original packet through the network path. This mitigates the impact of packet loss through the network, at the cost of increased bandwidth usage. Communication Manager will support the configuration of up to 3 redundant packets in Relay modes; Pass-Through modes are limited to no redundancy or 1 packet redundancy.

Table 3

Bandwidth Comparison Table				
	Redundancy = 0	**Redundancy = 1**	**Redundancy = 2**	**Redundancy = 3**
T.38 and Avaya Fax Relay				
30 ms payload	27.2 Kbps	54.4 Kbps	81.6 Kbps	108.8 Kbps
Fax/Modem Pass-Through				
10 ms payload	120.0 Kbps	240.0 Kbps	N/A	N/A
20 ms payload	92.0 Kbps	184.0 Kbps		
30 ms payload	82.7 Kbps	165.3 Kbps		
40 ms payload	78 Kbps	156 Kbps		
50 ms payload	75.2 Kbps	150.4 Kbps		
60 ms payload	73.3 Kbps	146.7 Kbps		
TTY-US, TTY-UK	Less than Codec	Less than Codec	Less than Codec	Less than Codec

Note — Less than Codec means less than 80 Kbps using no redundancy.

Since Pass-Through connections are treated essentially as audio calls, they use the same DSP (Digital Signal Processing) resources in a gateway media processor as a single G.711 connection. A Fax/Modem Relay, in comparison, requires the equivalent DSP resources of 4 G.711 calls (see Table 4). Therefore, Pass-Through is the preferred mode when conserving resources. There are 64 DSP resources per MedPro circuit pack, G700 and MM760. The G350 has only 32 DSP resources.

Table 4: DSP Resource Utilization by Call Type

Codec/Call Type	DSP Resources	Calls/DSP	
		w/o encryption	w/AES encryption
G.711 Pass-Through Clear Channel	1	16/DSP	12/DSP
G.723 G.729 Voice over IP	2	8/DSP	6/DSP
Fax Relay Modem Relay T.38	4	4/DSP	NA

Relay or Pass-Through, which should I use?

There is a trade off between bandwidth and DSP resources. Typical Fax relay uses 4 DSP resources but only ~27K of bandwidth. Pass-through uses 92K bandwidth but only 1 DSP resource. Table 3 displays these trade-offs. If bandwidth is a major consideration, relay may be the proper choice for your business. If DSP resources are constrained, pass-through may the proper choice. Where applicable, additional DSP resources applied to the solution provides for greater design flexibility.

Security

Avaya media encryption provides a secure solution for environments where enhanced voice privacy over a LAN/WAN is required. The communication device uses the 104-bit Avaya Encryption Algorithm or the Advanced Encryption Standard (AES) to encrypt voice packets. If media encryption is configured, the algorithm used during the audio channel setup of the call will be maintained for most fax and modem relay and pass-through modes (see Table 4). The exception is the T.38 Standard for Fax over IP; in this mode encryption will not be used. Note that encrypted calls utilize 33% more DSP capacity than non-encrypted calls (see table 4 above for details).

Table 5 Fax/Modem Encryption Support

Call Type	AEA (Avaya Encryption Algorithm)	AES (Advanced Encryption Standard)
Modem Pass-Through	Y	Y
Modem Relay	Y (After Release)	N (After Release)
Fax Pass-Through	Y	Y
Avaya Fax Relay	Y	N
T.38 Standard Fax Relay	N	N
TTY Relay	Y	Y
TTY Pass-Through	Y	Y

Interoperability

T.38 has emerged as the key standard for ensuring that VoIP capable networks can continue to support user requirements for real-time fax communications, and enables enterprises to support fax transmissions over IP in a multi-vendor environment. T.38 Fax relay is the ITU-T defined standard for Fax relay. It involves H.323/H.245 signaling capabilities exchange for Fax relay. Fax bearer traffic is relayed (encoded/decoded) according to the T.38 specification.

Avaya has incorporated this standard into its Media Processor TN2302BP circuit packs and VoIP media modules for media gateways so that they will interoperate with non-Avaya systems such as the Clarent or Alcatel Gatekeeper with Cisco and Clarent Gateways. CM systems can also use T.38 to transmit Faxes between other CM systems, Modular Messaging, and Multitech's MultiVoIP. Selection of T.38 for fax transmission involves CM negotiating with the far end system to ensure it will support T.38.

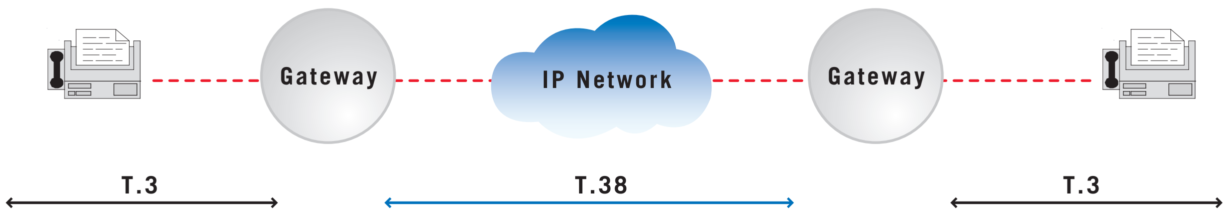
Avaya Media Gateways communicate with the fax devices using the standard ITU-T T.30 protocol used by all analog fax devices today. Using modem modulation, the sending fax machine sends T.30 protocol and fax image data to the media gateway, which demodulates the signals and repackages them into T.38 packets. The T.38 protocol provides an ITU-standard mechanism for a gateway/controller to inform another gateway of the desire to change the media stream from a voice stream to a data stream. Communication Manager provides this signaling to external T.38-enabled gateways to negotiate the session. The sending gateway then sends the T.38 packets to the receiving T.38-enabled gateway over IP (UDP), which then delivers the packets using the T.30 protocol to the endpoint fax device.

Multi-Tech models certified include:

- MVP 810-AV (Load 9.06.SI or higher)
- MVP 410-AV (Load 9.06.SI or higher)
- MVP 210-AV (Load 9.06.SI or higher)
- MVP 130-AV-FXS (Load 2.06.OT or higher)

Figure 2

T.38 Real-time (relay) system



Avaya's implementation of the ITU's T.38 standard is intended to interoperate with any T.38 compliant product. Remember this means relay mode only. There is a difference between compliance to a specification and certification from Avaya. Avaya has tested interoperability with Cisco version 2 gateways, Clarent gatekeeper, Alcatel gatekeeper, but has certified Mutli-Tech.

Requirements

Avaya Fax relay is supported by MV 1.3 and later

T.38 Standard Fax over IP support requires ACM 2.1.1 and the following VoIP Media Processors:

- TN2302AP hardware version 10 or higher with the latest firmware. This implies hardware that is less than version 10 will not support FoIP, MoIP or ToIP.
- MM760 VoIP Media Module v42 or the latest firmware
- G700 Media Gateway VoIP v42 & MGP 22.16 or the latest firmware
- G350 Media Gateway 22.16 or the latest firmware

Note: Modem and TTY have the same requirements as listed above. Also, the TN2602AP will not support F/M/ToIP until sometime in 2005.

Important Note: VoIP Media Processors are currently selected using a round-robin algorithm to balance the load. This will not change with the implementation of T.38. Therefore, all media processors in a network region administered for T.38 Fax/Modem over IP must be meet the above requirements.

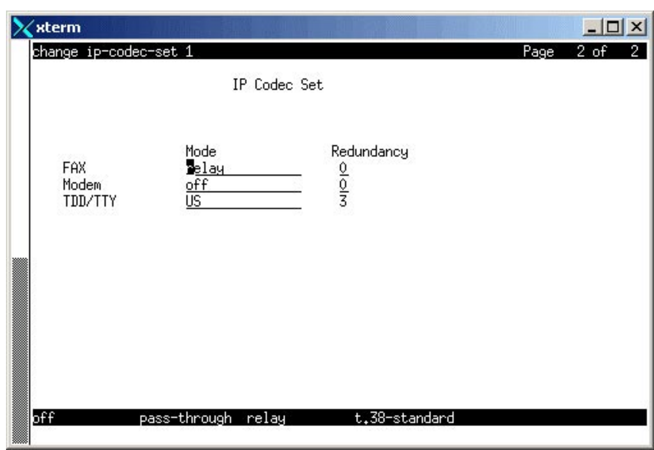
Administration

Avaya Fax Relay is a proprietary method of Fax transmission that is currently supported on older releases of CM (MV 1.3) and will continue to be supported for backwards compatibility in 2.1.1. This feature *cannot be administered* in releases prior to 2.1.1 though, and by default is always on. TTY (US & UK) also has no administration until ACM 2.1.1 and by default is always on.

The new enhancements for this feature in 2.1.1 are the SAT/ASA administration on "change ip-codec-set" form as well as the ability to administer redundancy for it. Also new in 2.1,1 is the ability of a 2.1.1 CM system set to use T.38 to automatically revert to use Fax Relay where possible if the far end Avaya system is older than CM 2.1 and therefore cannot support T.38. For T.38 and pass-thru modes to work properly the features must be correctly administered " change-ip-codec-set" form on both ends of the connection (i.e. setting both ends to the same mode of operation). Also, if the fax is sent between gateways or port networks on the same server, then the fax will be sent using Fax relay no matter how the fax mode was set.

Modem and TTY have similar settings in the IP-Codec-Set form beginning with ACM 2.1.1. Figure 3 shows the IP-Codec-Set form as an example.

Figure 3



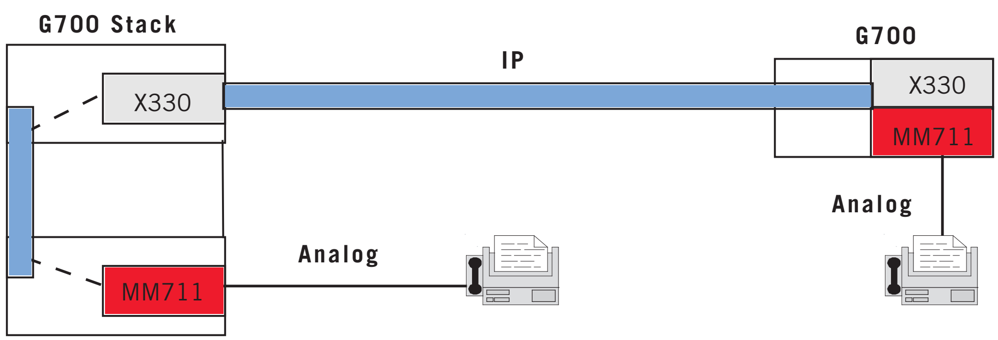
Configurations

If a network can support Fax or Modem Pass-Through, you will get higher throughput compared to either Standard or proprietary relay. Therefore Pass-Through may be the preferred method of implementing FoIP and MoIP in certain configurations.

Relay can work well, although slower, between gateways over an IP trunk. Beware of using multiple IP trunks for an end-to-end solution using fax relay. The transcoding delays may cause unpredictable results or tear down of the call.

Pass-Through can succeed where relay fails in this instance. One practical application is a stack of G700 gateways where the MM711 analog media module is located in one gateway and the IP trunk is located in another. The communication path is the Octaplane and stacking modules. The Octaplane system acts as an IP bus. Therefore, from the fax machine (or Modem) one transcoding has already been used to get to the gateway that has the IP trunk card. Going over the IP trunk is the second transcoding and the Fax or Modem call will most likely fail. See the diagram below.

Figure 3A — Not Supported for Relay



Notice the green areas in the preceding diagram that represent an IP path. Multiple transcodings will most likely prevent FoIP/MoIP from working or working well. This configuration is not supported.

Figure 3B — Supported for Relay

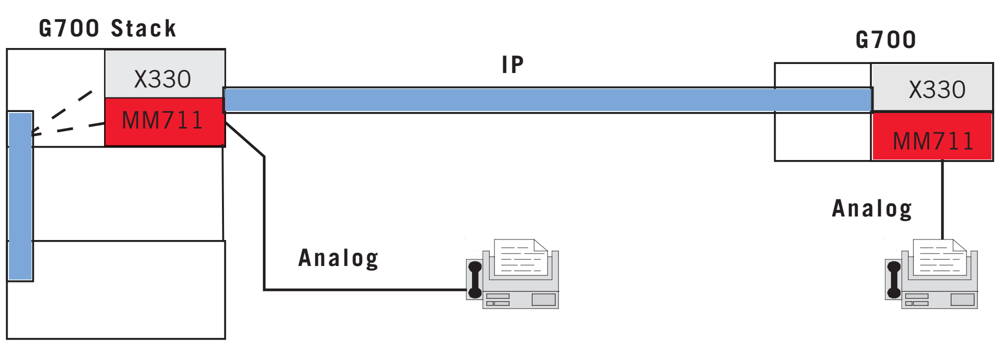


Figure 3B is supported for Relay, both Avaya proprietary and T.38 because there is only one transcoding from analog to IP. Both Figures 3A and 3B could work for Pass-Through at the cost of increased bandwidth.

Many configurations are supported and it would be difficult to list them all. The following diagram depicts many common implementations utilizing different servers and gateways as well as interconnecting network regions with trunks. There may be a sizable LAN infrastructure between the port networks or gateways even though they are not shown. This is a normal condition for the LANs in the office and enterprise.

Network Region 1 is an enterprise with an S8700 server as an IP connect system and a port network. The port networks can be MCC, G650 or G600 cabinets.

Network Region 2 is a G350 with two DS-1 trunk cards. The MM340 has an integrated CUS/DSU and the MM342 requires an external CSU/DSU.

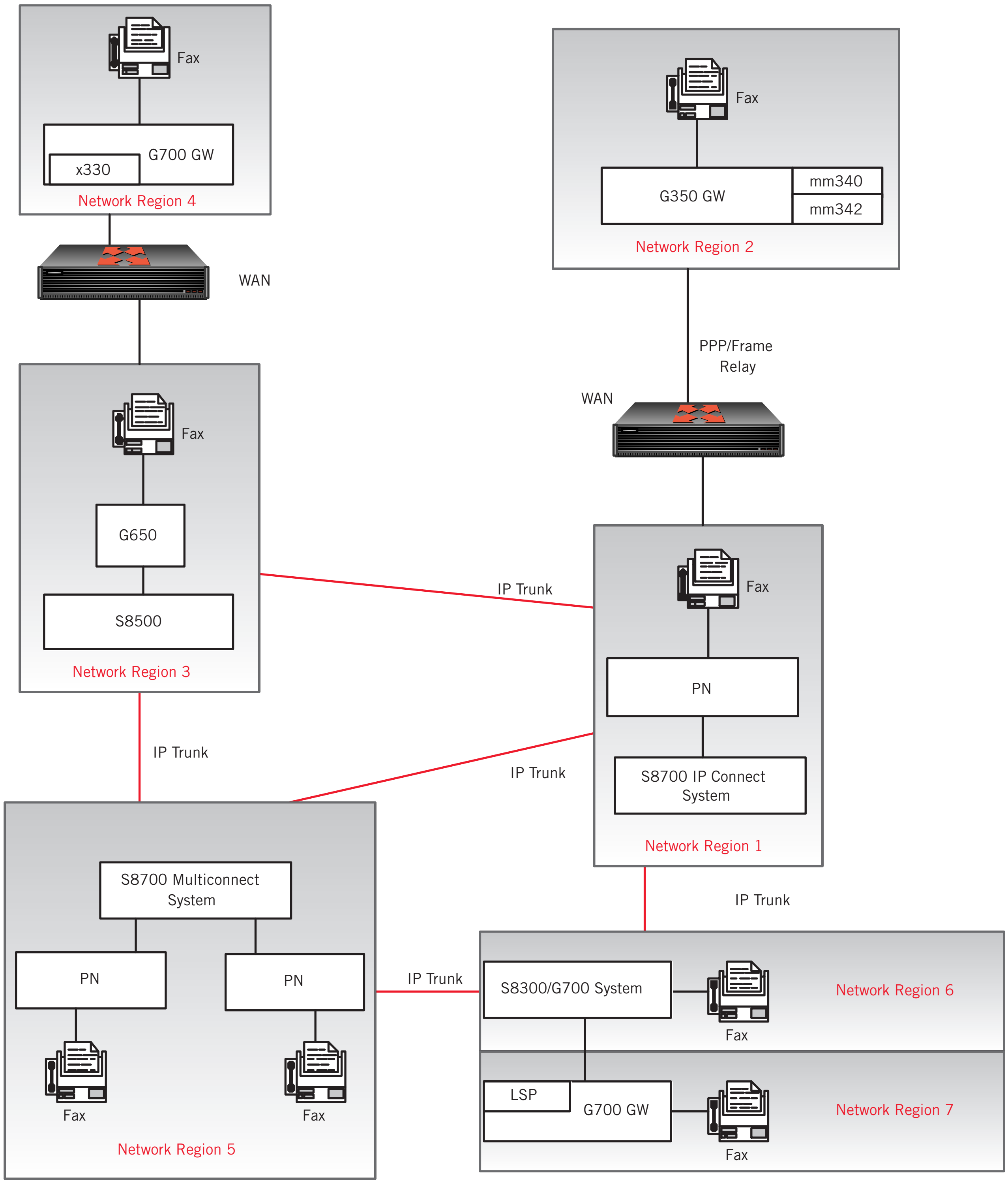
Network Region 3 is an S8500 server and G650 port networks.

Network Region 4 is a G700 with an X330-2DS1 expansion module.

Network Region 5 is an S8700 server as a multi-connect system and multiple port networks.

Network Region 6 is an S8300 server in a G700 gateway.

Network Region 7 is a G700 gateway with an LSP that is directly connected to the G700 in Network Region 6.



Notes:

1. T.38 is changed to Fax-relay for IGC calls.

FAQs

Question: Is fax relay supported now in 1.3 and 2.0 or only in 2.1.1?

Answer: Fax relay is supported in 1.3 and 2.0. It's proprietary solution with no Admin and, implemented entirely on the Medpros and gateway VoIPs. With this solution, Avaya can only fax to Avaya platforms (within G700s/G350s and Medpros). Think of this as Avaya's proprietary fax relay. New to 2.1.1 will be the support of the standard's based fax relay called: T.38. This is the ITU-T standard, which involves signaling and the ability to interoperate with other vendors that support T.38 fax relay. Admin on the 2nd page of the IP codec set form allows for the administration of the type of fax transport to be used over IP trunks:

1. "relay" (proprietary)
2. "T.38"
3. "fax pass-thru"
4. "OFF"

The default is "relay" and this is used on all IGC calls under the same controller.

Question: In a S8700 IP Connect scenario, which of the Modem features, Pass Through or Relay, would work for Inter-Port Network modem calls. Calls from the PSTN via a PRI are sent to another Port Network over the VoIP Inter-Port Network bearer path (Med Pro). In most situations, the modem calls will not handshake. Will the CM2.1.1 release help this?

Answer: Modem pass-thru or relay should work over Inter-port networks. Direct IP/shuffling must be enabled as in fax relay operation in order to limit the call to a single IP hop. Prior to this release, most of these modem calls failed because either the configured audio codec was used to transport the fax (not suitable due to compression and voice signal processing) or the call went into a fax relay mode thinking its was a fax call instead of modem (fax tones and modem tones are very similar and there was no distinguishing them in 2.0 or prior). So modem handshaking should correctly take place in the 2.1 load with these features.

Question: Will future ACM code support T38 Fax/Relay interoperability with IP Office?

Answer: IP Office has been making changes to interoperate with our existing Fax Relay solution (proprietary) rather than the new T.38. They may include some administrative options to give the customer the ability to admin IP Office style fax or ACM style fax with the Medpro/G700/G350. They do have plans to support T.38 in the future.

Question: I'm a technician and I've heard that Fax Pass-Through can be used running ACM2.1.0, but I can't administer it using my craft login. What's the deal?

Answer: Fax Pass-Through is officially supported in ACM 2.1.1, however, the code didn't change from ACM 2.1.0 to ACM 2.1.1. Fax Pass-Through can be administered using the "init" login for an ACM 2.1.0 system. The "craft" login *will* work in the later ACM 2.1.1. Also, please note that T.38 relay will NOT work in ACM 2.1.0 systems.

Question: I've heard fax relay isn't as robust as fax pass-through — is this true?

Answer: There have been issues for some customers where pass-through seemed to work better than relay. At this time, we believe the problems were all related to a lack of DSP resources under heavy faxing and calling loads. It is imperative that proper DSP resource planning is done before implementing fax relay. Remember, relay takes 4 DSP resources whereas pass-through takes only 1 DSP resource. The tradeoff is bandwidth. Relay takes little bandwidth compared to pass-through.

Questions and comments can be directly to afunguy@avaya.com

Appendix C

Change History:

Issue 1.0 October 2004 — Initial release

Issue 1.1 November 2004 — Expanded bandwidth table and clarified tradeoff between bandwidth and DSP resource usage. Also, feature release dates are more specific and added FAQs.

About Avaya

Avaya enables businesses to achieve superior results by designing, building and managing their communications networks. Over one million businesses worldwide, including more than 90 percent of the FORTUNE 500®, rely on Avaya solutions and services to enhance value, improve productivity and gain competitive advantage.

Focused on enterprises large to small, Avaya is a world leader in secure and reliable IP telephony systems, communications software applications and full life-cycle services. Driving the convergence of voice and data communications with business applications — and distinguished by comprehensive worldwide services — Avaya helps customers leverage existing and new networks to unlock value and enhance business performance.

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