Lexmark Fax over IP

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Abstract

This document discusses the benefits of the Lexmark Fax over IP solution. It is assumed that readers are familiar with topics associated with fax and Voice over IP.

Edition notice

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Introduction

As Voice over IP (VoIP) has taken over the world of telephony, there is still one function that requires the old analog phone lines: fax. Many business processes use fax due to its ubiquity, ease-of-use, and legal status. However, supporting analog fax lines and debugging fax issues on VoIP phone systems costs businesses much time and money.

This document describes how the Lexmark Fax over IP (FoIP) solution overcomes these problems. Lexmark's FoIP eliminates the need for a modem and analog lines by using T.38: the real-time Fax over IP network protocol of the ITU. T.38, combined with either SIP or H.323 (the two most common VoIP protocols), provides robust fax performance in a VoIP environment. See the *Further VoIP and FoIP Reading* section at the end of this document for more information on T.38, SIP, and H.323.

Typical Voice over IP Installation

In a typical VoIP installation, analog phone lines for fax are provided by an Analog Telephone Adapter (ATA). The ATA connects to the IP network; it detects when a fax call is starting and converts the analog tones and HDLC frames of a normal fax call into the T.38 network packets that can move along the IP network. The ATA is usually in a wiring closet connected to analog lines that run throughout the building. At the site of a given multifunction printer (MFP), an analog line connects to the modem on the MFP while a network cable connects to the MFP network adapter.

The Lexmark Fax over IP Solution

Lexmark's FoIP solution eliminates the need for the ATA, MFP modem, and analog wiring. The fax data is preformatted into the T.38 standard, and the MFP sends the T.38 packets directly to the IP PBX. Lexmark FoIP saves customers money and effort by removing the need to maintain this out-of-date equipment. It also makes for a much simpler connection to the IP PBX without any of the analog to digital conversions required in the typical installation. With FoIP, there is only one connection (network) from the wall to the MFP instead of two (phone and network). See Figure 2.

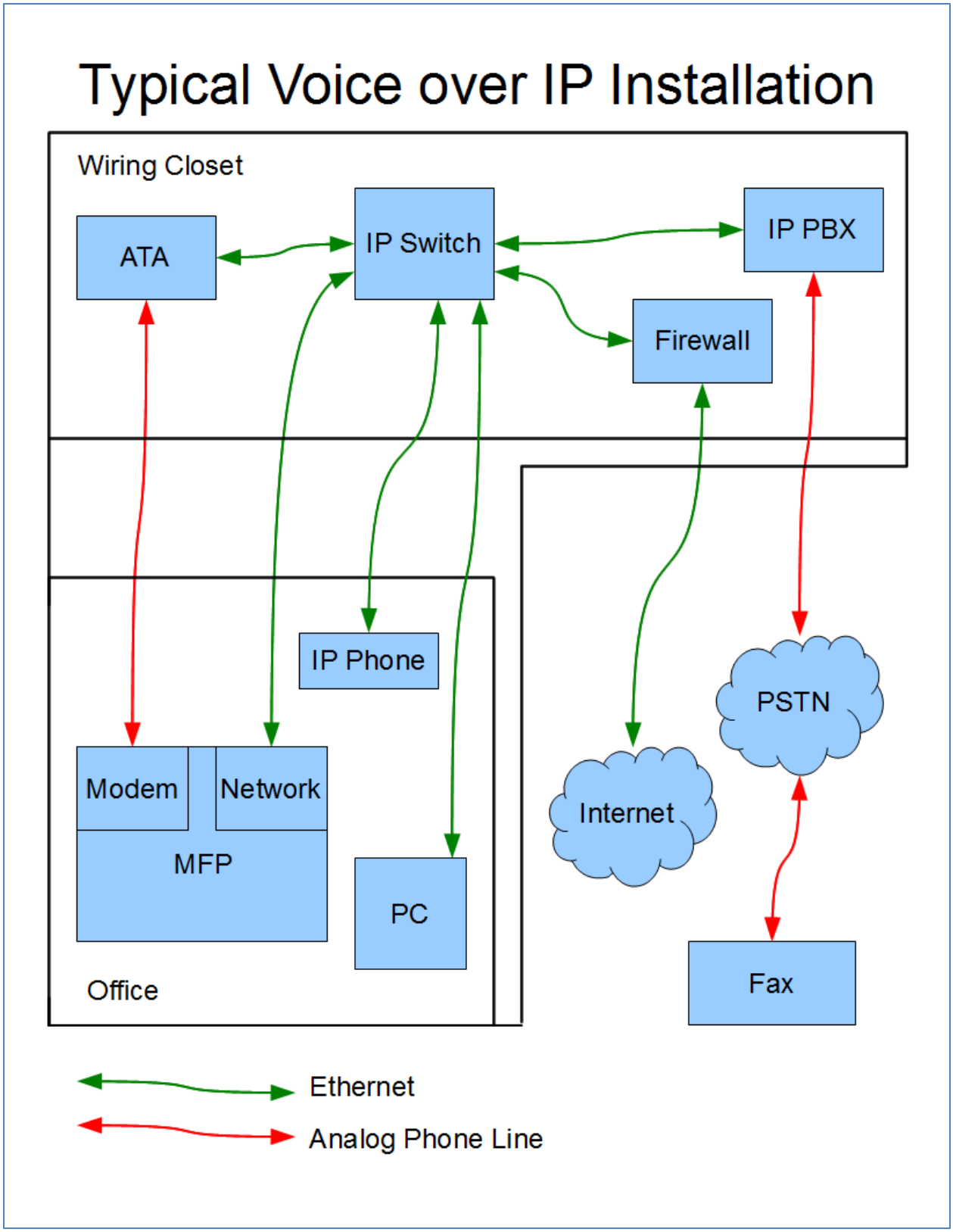

Typical Voice over IP Installation
Wiring Closet
ATA
IP Switch
IP PBX
Firewall
IP Phone
PSTN
Modem
Network
MFP
Internet
PC
Office
Fax
Ethernet
Analog Phone Line

Figure 1

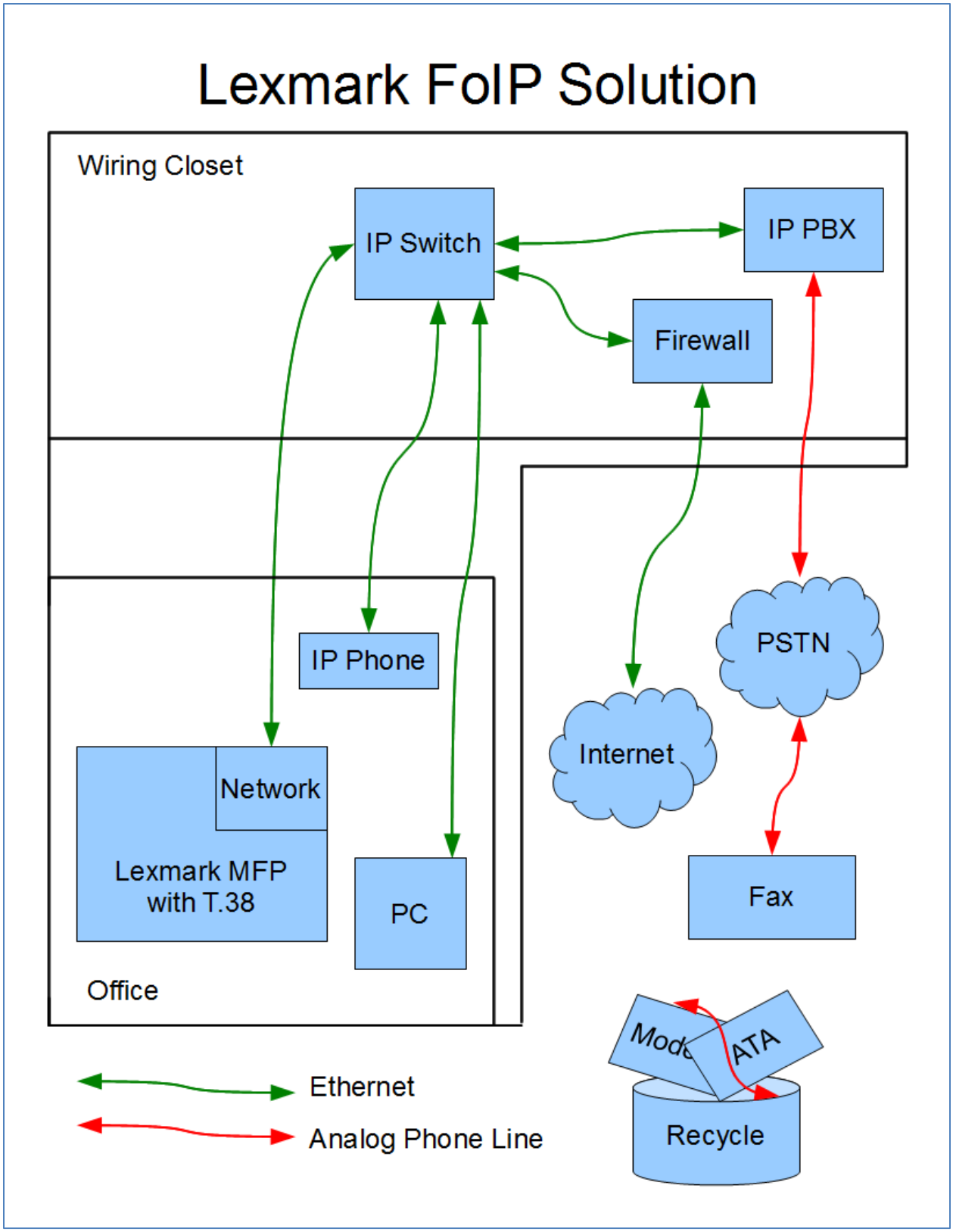

Lexmark FoIP Solution
Wiring Closet
IP Switch
IP PBX
Firewall
IP Phone
PSTN
Internet
Network
Lexmark MFP
with T.38
PC
Fax
Office
ATA
Recycle
Ethernet
Analog Phone Line

Figure 2

No Change for the End User

The Lexmark FoIP solution is designed so that there are no changes for the end user. The user experience for all fax-related functions—like sending and receiving faxes, Embedded Solutions Framework (eSF) applications, and fax forwarding to FTP, LDSS, and file shares—are the same for T.38 as they are for analog fax. The only exception is that users cannot turn the speaker on to hear the fax sounds because there are no sounds with T.38.

This identical experience is possible because of the architecture of the Lexmark Fax System, as seen in Figure 3. It is determined at the modem layer whether to use the analog modem or network with T.38. All layers of code above the modem layer do not know if the fax is being transmitted via analog modem or network. This removes the possibility of there being any difference in the handling of the fax data in other parts of the system.

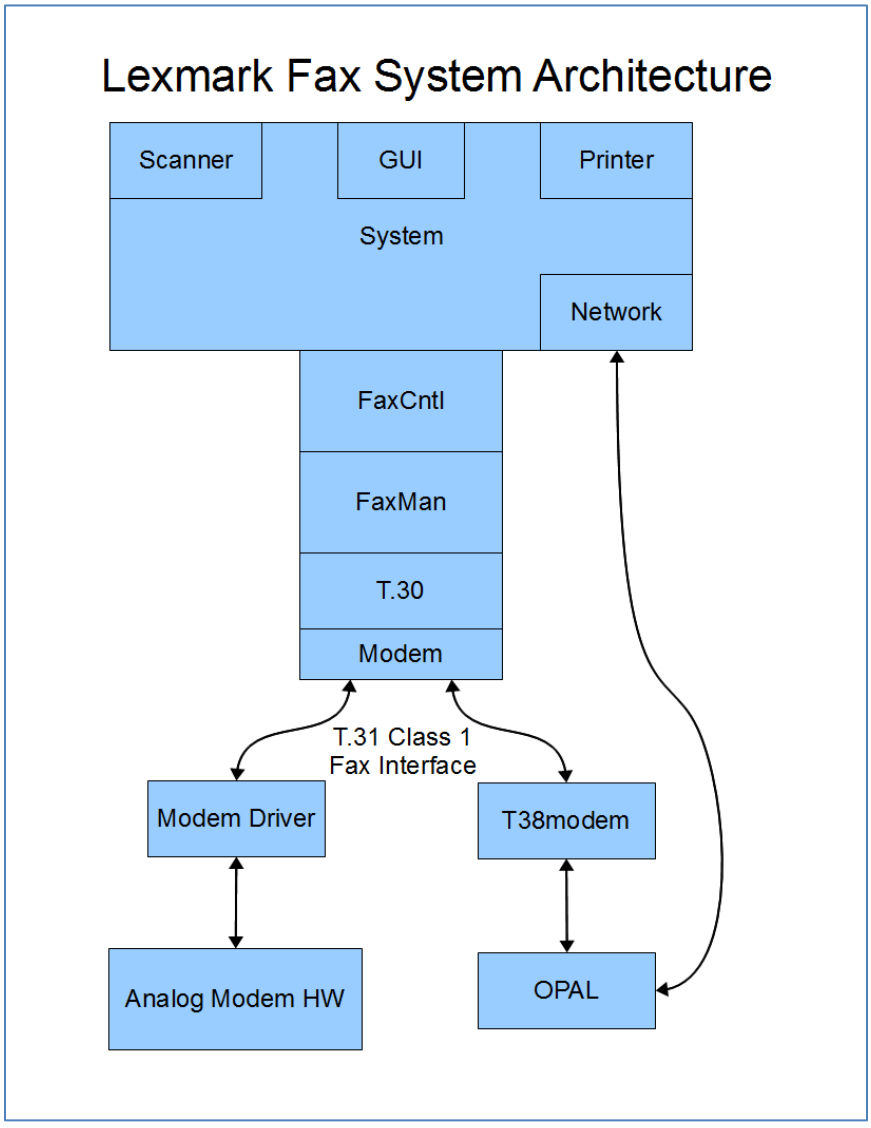


Figure 3

Reliability of the Lexmark FoIP Solution

The Lexmark FoIP solution uses two popular open source packages to provide the necessary protocols: Open Phone Abstraction Library (OPAL), for the SIP and H.323 VoIP protocols, and T38Modem, for the T.38 protocol. OPAL is a C++ multi-platform, multi-protocol library for Fax, Video, and Voice over IP as well as other networks. It is used in many products, like Ekiga: an open source softphone, video conferencing, and instant messenger application. T38Modem is an open source package that uses OPAL to access a VoIP phone system and then presents a T.31 Class 1 fax interface to the Lexmark Fax System. T38Modem is used in many products including the enterprise-class open source fax server Hylafax. Because these open source packages are backed by the experience of thousands of users and millions of faxes sent over the years, Lexmark's customers receive the benefit of a reliable product. See the further reading references at the end of this document for more information about OPAL and T38Modem. FoIP uses T38Modem 2.0.0.

Setup of the Lexmark FoIP Solution

Lexmark FoIP can be setup using the MFP control panel, Embedded Web Server, and UCF files. The setup is similar to the configuration of a SIP softphone on a PC. The following sections show the settings on the Embedded Web Server.

Fax Mode

In existing Lexmark products, Fax Mode selects between Analog Fax and Fax Server. Now, with T.38 support added, Fax Mode selects between Analog/T.38 Fax and Fax Server. See Figure 4.

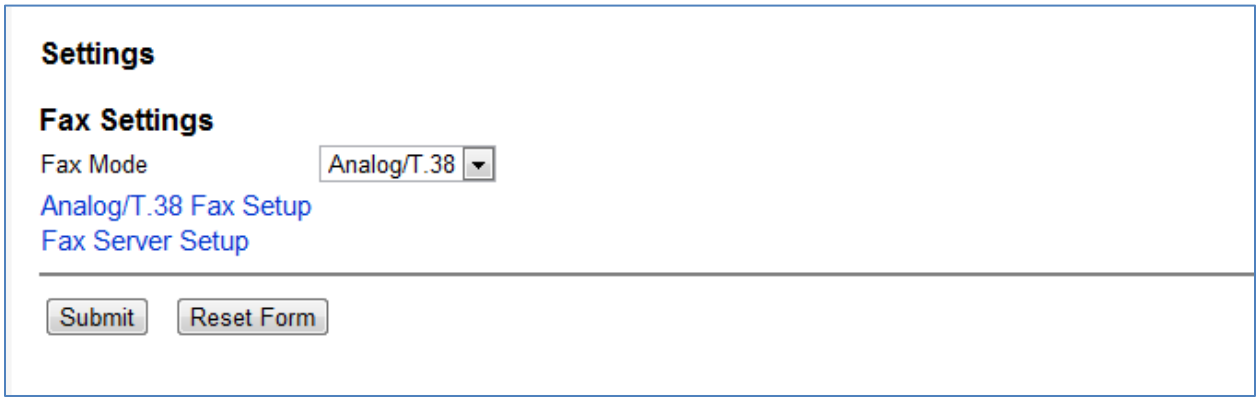


Figure 4

Fax Transport

Once Fax Mode has been set to Analog/T.38, Fax Transport is used to select between Analog Fax and T.38 Fax. See Figure 5.

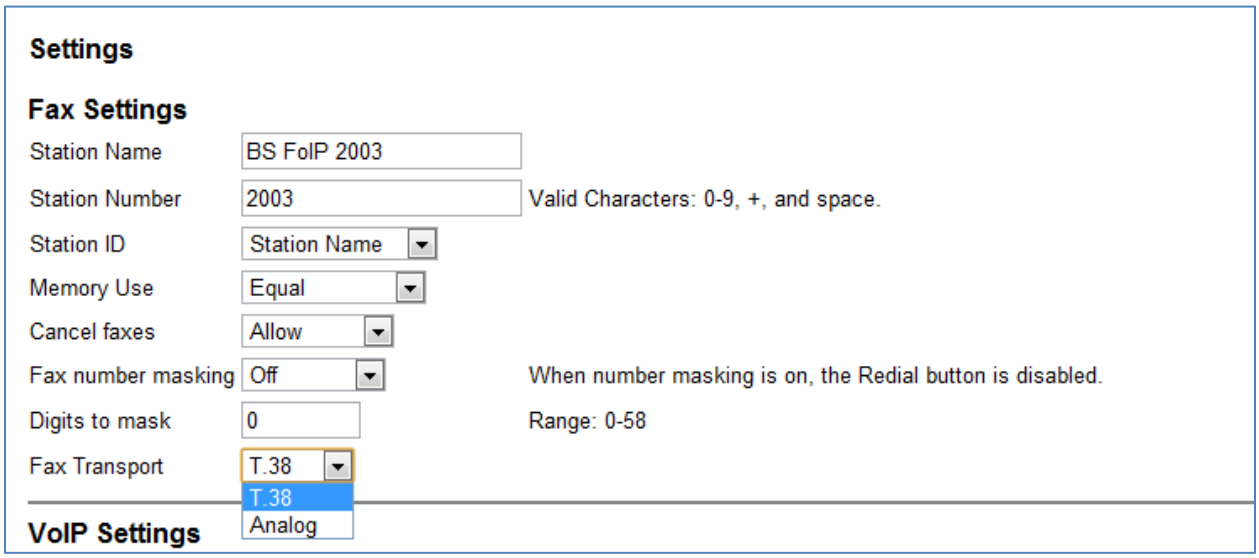


Figure 5

VoIP Settings

There is a new group of settings, described below, called VoIP Settings. See Figure 6.

- **VoIP Protocol:** This is used to select which VoIP protocol to use. The choices are SIP, H.323, and SIP and H.323. When SIP and H.323 is selected, outgoing calls will default to SIP unless a SIP Dial Out Digit is selected.
- **Trace Level:** This is used in debugging VoIP issues. The higher the number the more debug data is gathered. This data is stored on the fax data partition.
- **STUN Server:** A STUN Server is used to traverse a fire wall.
- **Force Fax Mode:** This is needed in some circumstances when the VoIP gateway does not automatically switch from analog to T.38 at the beginning of a fax call. Setting Force Fax Mode may help in these cases.



Figure 6

SIP Settings

The SIP Settings described below are used to configure the SIP protocol. See Figure 7.

- **Proxy:** This is the name or IP address of the system that converts the destination phone number to an IP address. In the example given in Figure 7, the Proxy is a Cisco UC520 system.
- **Registrar:** This is the name or IP address of the system that registers SIP clients. In the example, the Registrar is the same as the Proxy, which the substitution variable ${SIPProxy} is used to indicate. See the *Substitution Variables for VoIP, SIP, and H.323 Settings* section for a list of available substitution variables.
- **User:** This is the name of the user for SIP. In the example, the extension number is used.
- **Password:** This is used for registering with the SIP Registrar. In the example, the substitution variable ${StationNumber} is used as part of the password.
- **Contact:** This is the value to be used in the SIP Contact field.
- **Realm:** This is the name of the realm to be used in the SIP communication. It is used in the Address of Record (AOR) when registering. If it is omitted, then the name of the proxy is used in its place.
- **Auth ID:** This is used as the user name when registering. If it is omitted then the User field is used instead. In the example, the substitution variable ${SIPUser} is used to set the Auth ID the same as User.
- **Outbound Proxy:** If this is set, then all SIP communication is sent through this system.
- **Disable Incoming Calls:** If this is selected, then all SIP incoming calls are ignored.
- **SIP Dial Out Digit:** This allows for a prefix digit to be selected to indicate that a call should go out as a SIP call and not as an H.323 call.
- **SIP Registration Status:** This shows the status of the SIP Registration. The first line should be "200 OK," which is the SIP response when there are no issues. Anything else in the first line indicates a problem with the SIP registration. The second line of this box is the AOR. The third line is the time of the registration.

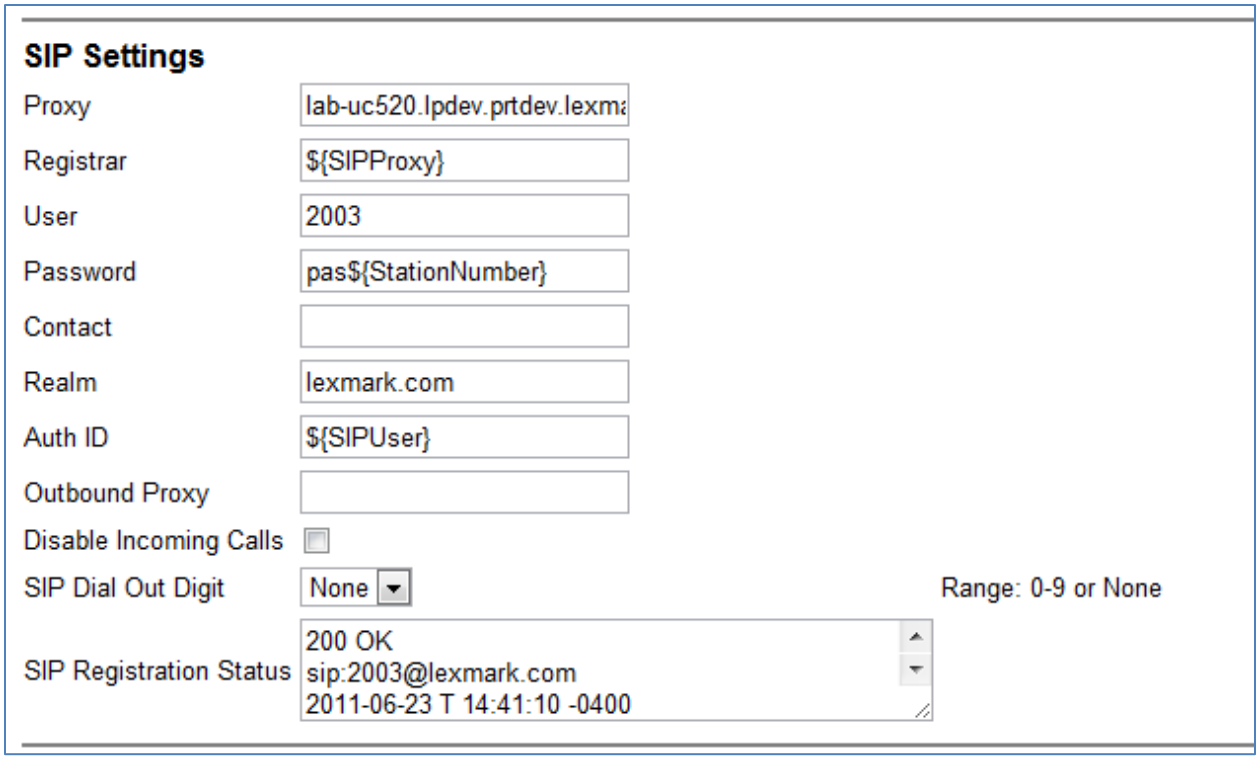

SIP Settings
Proxy
lab-uc520.lpdev.prtdev.lexma
Registrar
${SIPProxy}
User
2003
Password
pas${StationNumber}
Contact
Realm
lexmark.com
Auth ID
${SIPUser}
Outbound Proxy
Disable Incoming Calls
SIP Dial Out Digit
None
Range: 0-9 or None
SIP Registration Status
200 OK
sip:2003@lexmark.com
2011-06-23 T 14:41:10 -0400

Figure 7

H.323 Settings

The H.323 Settings are used to configure the H.323 protocol. See Figure 8.

- **Gateway:** This is the H.323 Gateway that is to be used.
- **Gatekeeper:** This is the H.323 Gatekeeper that is to be used.
- **User:** This is the User that is used in registering with the Gateway.
- **Password:** This is the Password that is used in registering with the Gateway.
- **Enable Fast Start:** If this is selected, then the H.323 fast start procedure is used.
- **Disable H.245 Tunneling:** If this is selected, then H.245 Tunneling is disabled.
- **Disable Gatekeeper Discovery:** If this is selected, then Gatekeeper discovery is disabled.
- **Disable Incoming Calls:** If this is selected, then all H.323 incoming calls are ignored.
- **H.323 Dial Out Digit:** This setting allows for a prefix digit to be selected to indicate that a call should go out as an H.323 call and not as a SIP call.

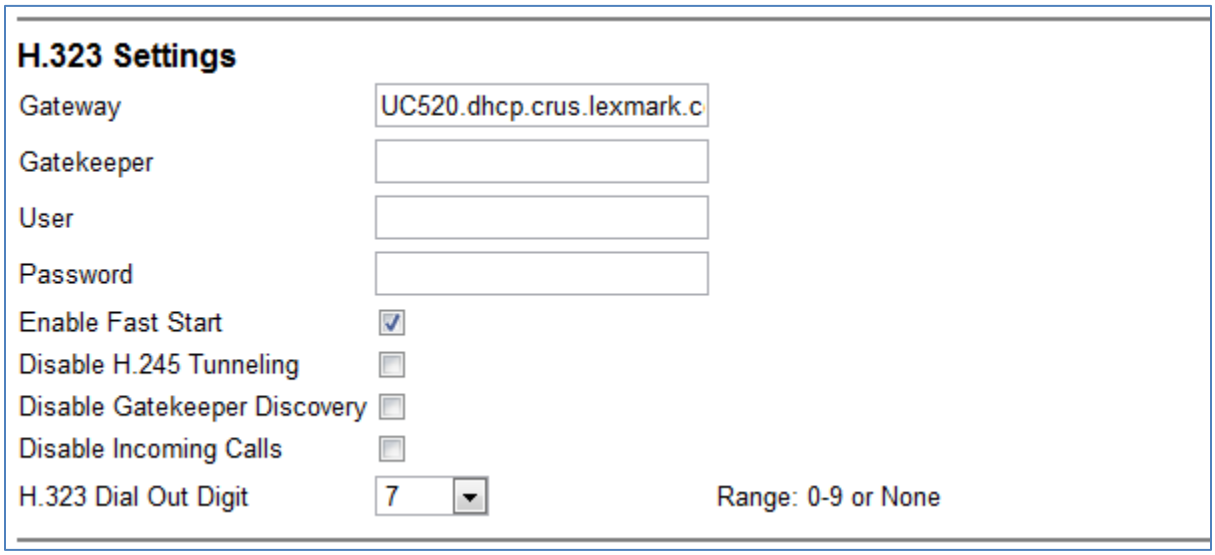


Figure 8

T.38 Settings

There is a new group of settings called T.38 Settings. These are used to configure the T.38 protocol. The first three are used to set the redundancy of the data transmitted to increase reliability. However, the enhanced reliability comes at the cost of extra bandwidth. Figure 9 shows the default settings.

- **Indicator Redundancy:** This sets the number of times that a fax indicator (CNG, ANS, etc.) is repeated in the T.38 communication.
- **Low Speed Redundancy:** This sets the number of times that low speed data (300 bps FSK signaling) is repeated in the T.38 communication.
- **High Speed Redundancy:** This sets the number of times that high speed data (image data) is repeated in the T.38 communication.
- **UDPTL Keep Alive Interval:** This sets the Keep Alive Interval for the UDPTL protocol.

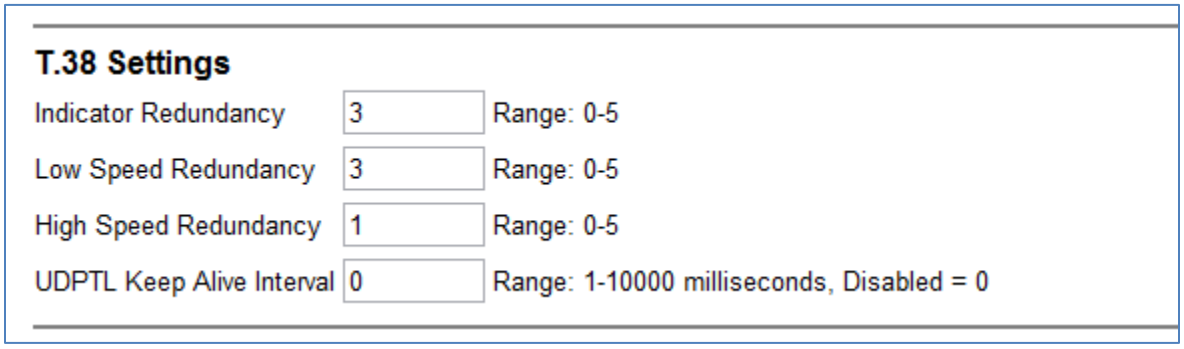


Figure 9

UCF

The following is a list of the UCF ID's for all of the related settings:

mfp.fax.mode	"0" for Analog/T.38 or "2" for Fax Server
mfp.fax.voipSettings.faxTransport	"0" for Analog or "1" for T.38
mfp.fax.voipSettings.voipProtocol	"1" for SIP, "2" for H.323, "3" for SIP and H.323
mfp.fax.voipSettings.traceLevel	"0" – "7"
mfp.fax.voipSettings.stunServer	"Name of STUN Server"
mfp.fax.voipSettings.forceFaxMode	"true" or "false"
mfp.fax.sipSettings.proxy	"Name of Proxy"
mfp.fax.sipSettings.outboundProxy	"Name of Outbound Proxy"
mfp.fax.sipSettings.registrar	"Name of Registrar"
mfp.fax.sipSettings.user	"Name of User"
mfp.fax.sipSettings.password	"Password"
mfp.fax.sipSettings.contact	"Contact"
mfp.fax.sipSettings.realm	"Realm"
mfp.fax.sipSettings.authID	"AuthID"
mfp.fax.sipSettings.disableIncomingCalls	"true" or "false"
mfp.fax.sipSettings.sipDialOutDigit	"-1" for None or "0" – "9"
mfp.fax.h323Settings.gateway	"Name of Gateway"
mfp.fax.h323Settings.enableFastStart	"true" or "false"
mfp.fax.h323Settings.disableH245Tunneling	"true" or "false"
mfp.fax.h323Settings.gatekeeper	"Name of Gatekeeper"
mfp.fax.h323Settings.user	"Name of User"
mfp.fax.h323Settings.password	"Password"
mfp.fax.h323Settings.disableGatekeeperDiscovery	"true" or "false"
mfp.fax.h323Settings.disableIncomingCalls	"true" or "false"
mfp.fax.h323Settings.h323DialOutDigit	"-1" for None or "0" – "9"
mfp.fax.t38Settings.indicatorRedundancy	"0" – "5"
mfp.fax.t38Settings.lowSpeedRedundancy	"0" – "5"
mfp.fax.t38Settings.highSpeedRedundancy	"0" – "5"
mfp.fax.t38Settings.udptlKeepAliveInterval	"0" for Disabled, "1" – "10000"

Substitution Variables for VoIP, SIP, and H.323 Settings

The following substitution variables can be used when configuring the VoIP, SIP, and H.323 settings:

${IPAddr}	The IP Address of the MFP
${SIPUser}	The User field in the SIP Settings section
${SIPProxy}	The Proxy field in the SIP Settings section
${SIPRegistrar}	The Registrar field in the SIP Settings section
${H323Gateway}	The Gateway field in the H.323 Settings section
${H323Gatekeeper}	The Gatekeeper field in the H.323 Settings section
${StationNumber}	The Station Number field in the Fax Settings section
${StationName}	The Station Name field in the Fax Settings section

Multiple indirection is acceptable. For example, SIP User can be set to ${StationName} and SIP Password to ${SIPUser}. However, avoid substitution loops: do not set SIP User to ${SIPUser}. This will cause the text "${SIPUser}" to be used for that field.

Also, substitution variables can be used as part of a field. For example, SIP Password could be set to pas${SIPUser}. In this case if the SIP User was set to 4321, then the SIP Password would be pas4321. Another useful example is to set the SIP Contact to ${SIPUser}@${IPAddr}.

On MFPs that have Fax Name instead of Station Name and Fax Number instead of Station Number, the original substitution variables ${StationName} and ${StationNumber} are still used.

Sending to an IP Address Instead of a Phone Number

The Lexmark FoIP solution also allows faxes to be sent to another T.38 fax machine at a specific IP address. When the phone number is in the form NNN*NNN*NNN*NNN, where each NNN is a number between 1 and 254, it is assumed to be an IP Address. The fax is then directed to that IP Address instead of the SIP Proxy or H.323 Gateway.

Note that setting the SIP Outbound Proxy will disable this feature, as all SIP communication will go to that Outbound Proxy.

Lexmark MFP Models with the Lexmark FoIP Solution

The Lexmark FoIP solution is currently available for the following Lexmark MFP models:

- X463, X464, X466
- X651, X652, X654, X656, X658
- X734, X736, X738
- X860, X862, X864
- X792
- X925
- X548
- X950, X952, X954

Compatibility with Avaya

Avaya H.323 is a proprietary solution; therefore Lexmark FoIP is not compatible with it. However, Avaya's SIP is not proprietary and works well with Lexmark FoIP.

Avaya systems only support fax speeds of 9600 bps and below. Avaya also forces ECM off. This eliminates color fax and good compressions, like MMR and JBIG, since these require ECM.

In the trunk group of Avaya Communications Manager, the field "Redirect on OPTIM Failure" must be changed from 5 seconds to the maximum 30 seconds. If this is not changed, then the MFP will not have enough time to answer the call before the timeout, and the call will fail. Without the change, the MFP can be made to work if the number of rings to answer is set to 1. However, with rings to answer set to 1, the caller ID information will be lost.

Compatibility with Cisco

There are no known culpability issues with Cisco VoIP systems.

The following is an example setup on a Cisco UC520 with two SIP extensions 2001 and 2002:

```
voice service voip
allow-connections sip to sip
 redirect ip2ip
 fax protocol t38 ls-redundancy 5 hs-redundancy 2 fallback none
 sip
  registrar server
  no update-callerid

voice register global
 mode cme
 source-address 192.168.10.1 port 5060
 max-dn 2
 max-pool 2
 authenticate register
 authenticate realm lexmark
 create profile

voice register dn  1
 number 2001
 no-reg

voice register dn  2
 number 2002
 no-reg

voice register pool  1
 id mac 0021.B790.FA63
 number 1 dn 1
 username 2001 password pas2001
 codec g711ulaw

voice register pool  2
 id mac 0021.B7D8.64AD
 number 1 dn 2
 username 2002 password pas2002
 codec g711ulaw
```

Further VoIP and FoIP Reading

- The ITU-T Recommendation T.38:
  - http://www.itu.int/rec/T-REC-T.38/e
- The ITU-T Recommendation H.323:
  - http://www.itu.int/rec/T-REC-H.323/en
- The IETF RFC 3261 SIP: Session Initiation Protocol:
  - http://www.ietf.org/rfc/rfc3261.txt
- Open Phone Abstraction Library (OPAL):
  - http://sourceforge.net/projects/opalvoip/
- T38Modem:
  - http://sourceforge.net/projects/t38modem/
- More information on both FoIP and VoIP:
  - http://www.soft-switch.org/foip.html
  - http://www.estoile.com/links/voip.htm
  - http://www.voip-info.org/wiki/view/FoIP
  - http://www.3cx.com/PBX/t38.html
  - http://www.allaboutfoip.com