KSZ8342 ATA Evaluation System

User's Guide

Version 1.1 / November 15, 2013

Purpose

Micrel has developed an ATA Evaluation System based on its KSZ8342 processor; this system consists of an ATA Universal board, which hosts the KSZ8342 processor, a set of daughter boards which work with the Universal board, and associated firmware and software. Together, these components form a functional ATA system and are intended for users to evaluate the KSZ8342 processor as well as a reference design for users to create their own solutions using the KSZ8342 processor.

This document describes the features of the firmware associated with the ATA Evaluation System; it further describes details steps on how to use and evaluate these features. In addition, it described how to update the firmware on the evaluation system.

Audience

This document is intended for developers and managers who are involved in product development using Micrel's KSZ8342 device. Before reviewing this document, it is highly recommended that the reader become familiar with Micrel's ATA Evaluation System and the following documents:

- KSZ8342 – ATA Evaluation System: Demo Guide
- KSZ8342 – ATA Evaluation System: Configuration User's Guide

In addition, the document assumes that the reader has working knowledge of:

- Networking – TCP/IP, DHCP, TFTP
- IP-based Telephony Protocols: SIP, RTP/RTCP
- IP-based Telephony call flows

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Glossary

ATA	Throughout this document, the phrase “ATA” will refer to the Micrel Universal ATA Board.
ATA-FXSO	This refers to the board combination of the Micrel Eval board (KSZ8342 ATA Universal Board – REV 2.0) plus the SiLabs SLIC daughter board (Si3217xFB-EVB Rev 2.1 with Si32919 & Si32178) which provides an FXS port and an FXO port.
ATA-2FXS	This refers to the board combination of the Micrel Eval board (KSZ8342 ATA Universal Evaluation Board – REV 2.0) plus the SiLabs SLIC daughter board (Si32260PB20SL2-EVB Rev 1.0) which provides two FXS ports.
ATA-1FXS	This refers to the board combination of the Micrel Eval board (KSZ8342Q ATA Universal Evaluation Board – REV 2.0) plus the SiLabs SLIC daughter board (Si3217xCFB10SL0-EVB Rev 1.0 with Si32171) which provides one FXS port.

1 Introduction

Micrel has developed an ATA Evaluation System based on its KSZ8342 VoIP processor; this system consists of an ATA Universal board, which hosts the KSZ8342 processor, a set of daughter boards which work with the Universal board, and associated firmware and software. Together, these components form a functional ATA system and are intended for users to evaluate the KSZ8342 processor as well as a reference design for users to create their own solutions using the KSZ8342 processor.

The universal board works with three types of daughter cards:

- 1 FXS port
- 2 FXS ports
- 1 FXS + 1 FXO port

This document provides information for users to evaluate Micrel's KSZ8342Q processor in conjunction with its associated firmware and software. The KSZ8342 processor itself is described in detail in its chip Data Book. To evaluate the processor and the firmware, Micrel has developed a fully functional ATA board to enable users to exercise the features of the firmware and software. This document describes:

- Features of the firmware and software
- Procedures to evaluate these features

2 Organization

The document is organized as follows:

Section 3: Contents of the ATA Evaluation System

Section 4: Quick Start

Section 5: Summary of ATA Features: FXS

Section 6: Summary of ATA Features: FXO

Section 7: Using ATA Features: FXS

Section 8: Using ATA Features: FXO

Section 9: ATA Universal Board Description

Section 10: ATA SLIC Daughter Boards Description

Section 11: Firmware: Build and Update

3 Contents of the Evaluation System

The ATA Evaluation Systems consist of several components:

- Hardware
- Firmware
- Documentation

These are described below.

3.1 Hardware

The hardware component of the ATA Evaluation System consists of:

- ATA Universal Evaluation Board (see Figure 3 for further details).
- ATA SLIC daughter board (see Figure 4 for further details)
- Power module

3.2 Firmware

The firmware component of the ATA Evaluation System consists of

- Boot loader
- Kernel + Application Image
- Source code of kernel and build tool chain

See Section 11, Firmware: Build and Update, for further details.

3.3 Documentation

The documentation component of the ATA Evaluation System consists of

- Board related

- Evaluation Board Schematic
- Evaluation Board PCB files
- Evaluation Board BOM
- KSZ8342Q IBIS files

- ATA Evaluation System: Demo Guide
- ATA Evaluation System: Configuration User's Guide

3.4 Additional Requirements

To use the Universal ATA evaluation board kit, the following items, **which are not supplied with the kit**, are required;

- Qty 1 or 2 – POTS telephones to connect to the RJ11 connectors on the Evaluation Board
- Qty 1 – 8-port Ethernet switch
- Qty 1 or 2 – VoIP telephone to connect up to the Ethernet switch (Recommend Cisco Model SPA303)
- Qty 1 – Laptop or desktop computer (Windows XP or Windows 7) to connect to the Ethernet switch. This can be done via a Trial install.
- Your favorite communication program such as TeraTerm or Hyperterminal.
- SolarWinds Trivial FTP Program, Available at: SolarWinds Trivial FTP

3.5 Access of Firmware and Documentation

The firmware and documentation is available via Github (http://github.com/). To access the firmware and the documentation:

- Have your Github account ready. In case you do not have one, create a Github account as follows:
    - click on the Github link above
    - select user name and password
    - follow instructions from Github

- If you have an NDA with Micrel, you can request Micrel to provide you with access – email requests to voip-support@micrel.com
- For an organization, Micrel can create a team access. In case you have multiple users, please provide the Github user names for all persons requesting access.

4 Quick Start

This section describes how to set up your environment to start testing the features of the ATA Evaluation System. This set up requires a LAN to be setup at the customer site. This section describes specific equipment needed for this LAN, how to connect them, and certain required configurations.

IMPORTANT: These steps below are also described in detail in the ATA Evaluation System – Demo Guide; please review that document and follow the steps described therein.

4.1 Equipment List

This section lists all **items that need to be setup prior to setting up the ATA Demo**. They are as follows (see Figure 1):

1) Local Area Network (LAN)
2) DHCP Server available on the LAN
3) Ethernet Switch with minimum 5 ports, and connected to the Router through LAN
4) Windows-based PC connected to the Ethernet Switch
    a) The PC must have a serial port.
    b) Terminal Emulator Program such as TeraTerm, Putty, or HyperTerminal.
5) Router, which supports ALG (Application Level Gateway), and is connected to the Internet. It must be configured as follows:
    a) The following port numbers must be enabled
        i) 5060: to allow SIP traffic
        ii) 8000: to allow RTP traffic
        iii) 8001: to allow RTCP traffic
    b) Enable SIP ALG
6) PBX Switching System
7) Two IP Phones: Cisco SPA303 (recommended) or any other IP Phones

8) Three Analog Phones with Caller ID Capabilities that will be plugged into the Micrel's ATA-FXSO and ATA-2FXS Boards

9) An external phone (Cell phone or other phone).

10) Ethernet Cables (CAT5e)

4.2 Connection and Verification Procedure

1) Connect all items as shown in Figure 1 below. Please contact your local IT department for configuration of the Router.

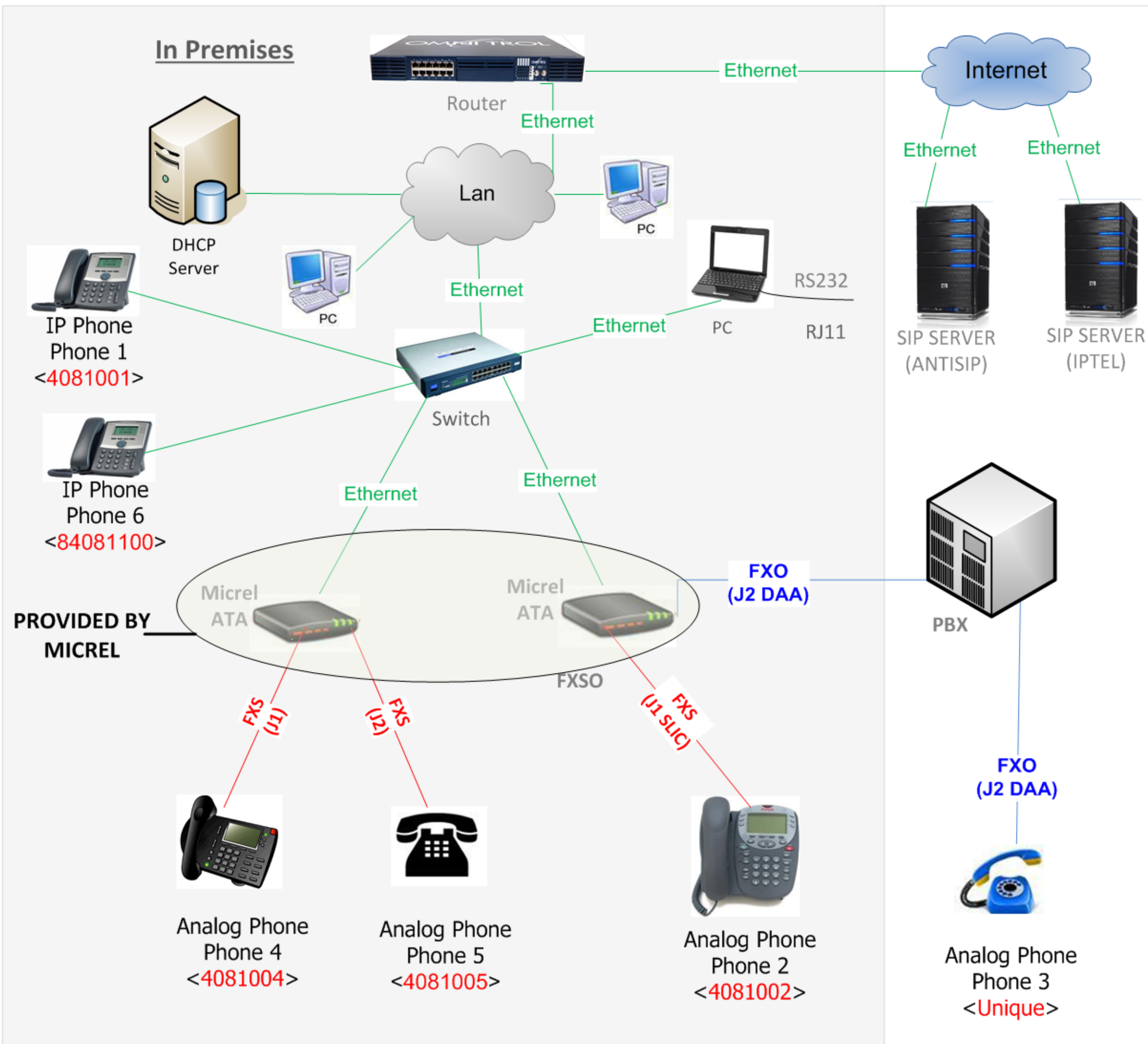

Figure 1: Setup required for ATA evaluation

2) Verify Network Connection:

   a) Make sure that the PC is setup for “Get IP Address Automatically”; in other words it must get the IP address from the DHCP server on the LAN.

   b) Check that the PC has obtained an IP address from the LAN; this verifies that the DHCP server is working.

   c) From the PC, ping the AntiSip Server by opening a command window and typing “ping sip.antisip.com” on the command line interface. This SIP Server should respond back. This ensures that the PC and other equipment connected to the Ethernet Switch will be able to contact the SIP Server.

3) Create Accounts on Antisip Server

   a) Micrel has set up accounts on an Internet based SIP server: Antisip (www.antisip.com). Micrel has created ten accounts on Antisip for use by customers. The account numbers are 4081000 thru 4081009. Contact Micrel Support (voip-support@micrel.com) to schedule the use of these extensions. If you would like to create your own accounts on Antisip, please refer to the ATA Evaluation System: Demo Guide for further information.

4) Create Accounts on IPtel.org SIP Server

   a) Micrel has set up accounts on an Internet based SIP server: www.iptel.org. Micrel has created two accounts on iptel.org for use by customers. The account numbers are 84081100 and 84081101. Contact Micrel Support (voip-support@micrel.com) to schedule the use of these extensions. If you would like to create your own accounts on IPTel, please refer to the ATA Evaluation System: Demo Guide for further information.

5) Verify PBX:

   a) To verify, plug an analog phone into the wall phone jack and check for dial tone. Also please note any prefixes that may be needed to dial out from the PBX Switch.

6) Configure 1st IP Phone (Phone1):

   a) The recommended IP Phone to use for this demo is the Cisco model SPA303. Before using the phone in this demo, it must be programmed. Follow the steps listed below to accomplish this programming. Refer to the Cisco SPA303 technical manual for any additional information.

   b) Get the assigned IP Address

i) On the IP Phone keypad, press the Menu key. This is the ▢ button below the “Envelope” button.

ii) Press down the key until “9 Network” is displayed and press Select

iii) IP Address is under “Current IP”. **Make a note** of it as you will use it in the next step.

c) Configure Antisip Server information on Phone1 using the Web GUI

i) Open the Web Browser on the PC and type the IP Address assigned to the IP Phone.

ii) Click on Admin Login (top Right corner), then on “Ext1” tab. (This phone happens to have 3 extensions)

iii) Change or confirm the following settings:

(1) Proxy: sip.antisip.com

(2) Display Name: Phone1

(3) User ID: 4081001

(4) Password: 4081001

d) Press “Submit All Changes” and wait about one minute for the phone to reboot.

In case you are using an IP Phone other than Cisco SPA303, consult its user’s manual to configure the phone as above.

7) Configure 2nd IP Phone (Phone6) – only for two SIP server scenario:

a) The recommended IP Phone to use for this demo is the Cisco model SPA303. Before using the phone in this demo, it must be programmed. Follow the steps listed below to accomplish this programming. Refer to the Cisco SPA303 technical manual for any additional information.

b) Get the assigned IP Address

i) On the IP Phone keypad, press the Menu key. This is the ▢ button below the “Envelope” button.

ii) Press down the key until “9 Network” is displayed and press Select

iii) IP Address is under “Current IP”. **Make a note** of it as you will use it in the next step.

c) Configure Antisip Server information on Phone1 using the Web GUI

i) Open the Web Browser on the PC and type the IP Address assigned to the IP Phone.

ii) Click on Admin Login (top Right corner), then on “Ext1” tab. (This phone happens to have 3 extensions)

iii) Change or confirm the following settings:

(1) Proxy: iptel.org

(2) Display Name: Phone6

(3) User ID: 84081100

(4) Password: 84081100

d) Press “Submit All Changes” and **wait about one minute** for the phone to reboot.

In case you are using an IP Phone other than Cisco SPA303, consult its user’s manual to configure the phone as above.

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5 ATA Features: FXS

The firmware provided as part of the ATA Evaluation System enables a rich set of VoIP telephony features. This section provides an overview of these features. The following section will provide details of how to use all these features.

5.1 Basic Services

- Call
  - Incoming / Outgoing
- Mid-call
  - Put call on Hold / retrieve call on Hold
  - Mute / Unmute
  - Second call
    - Make second call
    - Switch between two calls
  - Conference call
    - 3-Way Call Conference with Third Party Manual Join
    - 3-Way Call Conference with Third Party Auto Join
  - Call waiting
  - Call Transfers
    - Attended Call Transfer
    - Normal Call Transfer
    - Auto Attended Transfer
  - Call Forwarding
    - On No Answer (CFONA)
    - Unconditional (CFU)

        - On Busy (CFOB)
    - Call back on busy (CBOB)
- Pre-call
    - Call pick up
    - Do Not Disturb (DND)
    - Abbreviated dialing
    - Absent service

5.2 Supplementary Services

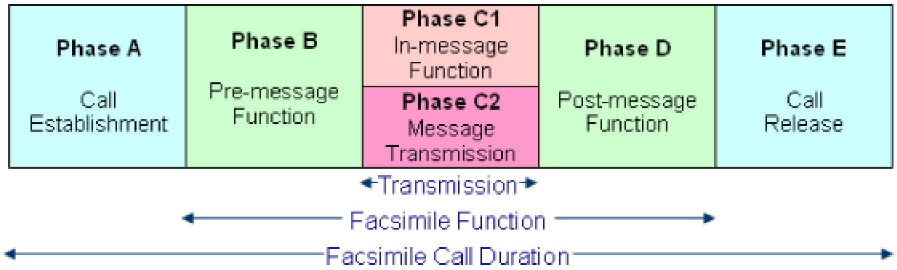
- Anonymous Call Blocking
- Caller Id Blocking
- Remote Caller Id Presentation
- Group Calling
- Hot Line/Warm Line
- Message Waiting Indicator (MWI)
- Music On Hold
- Outgoing Call Baring
- Secretary Service
- Volume Up/Down
- Wakeup Call Service

5.3 FAX Services

There are two modes of FAX operation:

- Fax Pass Through mode

- T.38 Fax mode



5.4 Features Supported via Keypad

This section summarizes all the features that are supported via keypad. They are described in detail in the following sections.

Table 1: List of Features Supported by Keypad

Feature Name	Feature Code	Parameter Description	Example
Conference	*46	-	*46
Conference (using JOIN header)	*74<UA1_number>	UA1_number is the remote user number (6010) where to join for conference	*746010
Normal Transfer	*49 <UA3_number>	UA3_number (6001) is the number where call is to be transferred	*496001
Attended Transfer	*45	-	*45
Mute/Unmute	*43<0/1>	'1' to enable mute and '0' to disable mute	*431 *430
Register/Unregister	*47<0/1>	'1' to enable register and '0' to disable register	*471 *470

Feature Name	Feature Code	Parameter Description	Example
Do not disturb (DND)	*44<0/1>	‘1’ to enable DND and ‘0’ to disable DND	*441 *440
Outgoing call barring	*48<0/1>	‘1’ to enable outgoing call barring and ‘0’ to disable outgoing call barring	*481 *480
Designated pickup (call pickup)	*67<UA2_number>	UA2_number (6002) is the number (ringing) whose call is to be received.	*676002
Call back on busy	*59	-	*59
Volume increase	*72	-	*72
Volume decrease	*73	-	*73

6 ATA Features: FXO

In addition to traditional ATAs that support FXS lines, one of Micrel's evaluation boards (ATA-FXSO) also includes and FXO port to support failover. This port is used in case the VoIP endpoint functions are not available on the FXS port – for example, the network is down, or the SIP server does not respond.

For the ATA-FXSO board there are three additional items to understand:

1) If you use 9 as the dial prefix, you get access to your PBX line (where the FXO is connected to). This way you can use your Phone connected to the FXS port simply as a regular analog phone; it will support all the features that your PBX supports, including FAX, call waiting, etc.

2) Another important aspect is failover – the FXS phone is configured for failover as follows:

   a) If you use SIP server 1 (using prefix "1") and the server does not respond, the system fails over to the FXO port and connects to the number dialed via the FXO line.

   b) Same as above for SIP server 2 (using prefix "2").

3) There are three additional configuration items provided

   a) Connect incoming FXO call directly to local FXS. If enabled, incoming calls on the FXO port are routed to the FXS line, similar to a regular analog phone.

   b) Play voice message when there is incoming call on FXO line (this is applicable when configuration item described in above is disabled). This is enable or disable flag. If enabled, it will play message otherwise dial tone will be played out, to indicate that caller needs to dial VoIP number.

   c) Play voice message when there is incoming VoIP call on FXO port. If enabled, it will play message otherwise dial tone will be played out, to indicate that caller needs to dial PSTN / PBX number.

7 Using ATA Features: FXS

7.1 Overview

This section describes in detail the procedures involved in exercising various features supported on FXS based ATA solutions. Evaluating the features requires one or more phone: some are analog phone that will be attached to the FXS port of the ATA board; others are SIP based IP Phones.

In SIP terminology, a phone is a User Agent (UA), which is a logical network end-point used to create or receive SIP messages and thereby manages a SIP session. A SIP UA can perform the role of a *User Agent Client* (UAC), which sends SIP requests, and the *User Agent Server* (UAS), which receives the requests and returns a SIP response. These roles of UAC and UAS only last for the duration of a SIP transaction. Based on this definition, the following additional terminology is used in this document:

- UA – Board: refers to Analog phones attached to the ATA board; essentially, it is the phone/FXS port being evaluated.
- UA – SIP phone is any device that supported VoIP telephony using SIP, such as:
  - another analog phone attached to the FXS port of the ATA board
  - another IP Phone or ATA
  - software based SIP Phone.
- SIP URI – This is the phone number assigned to a SIP phone and registered with a SIP server.
- FLASH – Refers to the Flash button on analog phone or rapid Off-Hook to On-Hook and On-Hook to Off-Hook transition.

7.2 Basic Services

7.2.1 Make Outgoing Call

User Agents: UA1 - Board, UA2 – Any SIP Phone

1) Assuming that UA1 and UA2 are registered to Proxy.
2) Once registration is through, go OFFHOOK on UA1's handset and dial the username of UA2 followed by # (pressing # is optional) eg. 888# or dial 888 and wait for inter-digit timeout.

3) UA2's handset indicates the incoming call by playing the ringing tone.

4) Go OFFHOOK on UA2's handset to accept the call.

5) The call is established.

7.2.2 Receive Incoming Call

User Agents: UA1 - Board, UA2 – Any SIP Phone

1) From UA2 dial UA1's SIP URI.

2) Incoming call is indicated at UA1 by the ringing of the handset.

3) Go OFFHOOK at UA1's to accept the call.

4) The call is established.

7.2.3 Terminating the Call

User Agents: UA1-Board, UA2-Any SIP Phone

1) When the call is in established state, go ON-HOOK to terminate the call. Any UA, either UA1 or UA2 can terminate this call by going ON-HOOK.

2) In case remote terminates the call, UA1 will hear busy tone.

7.2.4 Putting Established Call On Hold

User Agents: UA1-Board, UA2-Any SIP Phone

1) When the call is in established state, at UA1, FLASH should be pressed to put the other end on hold. Any UA, either UA1 or UA2 can put the call on hold.

2) Alternatively call can be put on hold by pressing the Hook to On-Hook and releasing it quickly. But this is not reliable.

Note that the call can be in hold for a configured timeout period.

7.2.5 Retrieving Held Call

User Agents: UA1-Board, UA2-Any SIP Phone

1) When the UA1 has put UA2 on hold, UA1 may press FLASH again to retrieve this call. The call will be established.
2) The UA that puts the call on hold, can retrieve the call.

7.2.6 Making Second Call

Making 2 simultaneous calls from same analog phone is supported. This section explains how to make the second call.

User Agents: UA1-Board, UA2-Any SIP Phone, UA3-Any SIP Phone

1) UA1 is in call with UA2.
2) UA1 should put UA2 on hold by pressing FLASH.
3) Dial tone is played at UA1.
4) Dial UA3's username followed by # (optional) from UA1's handset.
5) If UA3 accepts this incoming call, then UA1 and UA3 will be in call.

7.2.7 Switching between Calls

When there are 2 calls, typically one of them is on hold. This section explains how to switch between the two calls. This procedure ensures that one call is always on hold.

User Agents: UA1-Board, UA2 - Any SIP Phone, UA3-Any SIP Phone

1) UA1 is in call with UA2 and UA3 (Procedure explained above). UA2's call is held by UA1 and UA3 is talking to UA1.
2) UA1 can switch between these two calls by pressing FLASH.

7.2.8 Call Waiting

User Agents: UA1-Board, UA2-Any SIP Phone and UA3-Any SIP Phone

1) Go ONHOOK on both the ATAs connected to the board.
2) Enable call waiting in the user features section of the web interface at UA1, if necessary.
3) Establish call between UA1 and UA2.
4) Call UA1 from UA3. The call waiting tone can be heard at UA1.
5) Press FLASH at UA1 to accept this incoming call from UA3.
6) UA1 and UA3 will in call and UA2's call will be held
7) Switching between calls can be done at UA1 by pressing FLASH as mentioned above.

7.2.9 Conference

This procedure establishes 3-way conference between UA1, UA2 and UA3. In the example below, UA1 mixes the streams received from UA2 and UA3.

User Agents: UA1-Board, UA2-Any SIP Phone and UA3-Any SIP Phone

1) Go ONHOOK on both the ATAs connected to the board
2) Enable conference in the user features section of the web interface at UA1, if necessary
3) Establish call between UA1 and UA2.
4) To include UA3 in conference, UA1 puts UA2 on hold by pressing FLASH.
5) UA1 dials UA3.
6) Once UA1 and UA3 are connected, UA1 presses key sequence "*46#" to start the conference.
7) Now UA1, UA2 and UA3 are in conference.

7.2.10 Auto Join

This procedure enables third UA to join into an on-going conversation.

User Agents: UA1-Board, UA2-Board and UA3-Board

Enable mode.

1) Enable Auto Join option on UA1 and UA3.
2) UA1 and UA2 are in call.
3) UA3 dial “*74<UA1 number>” to join the call UA1 and UA2 are having.
4) UA1, UA2 and UA3 are in conference call.

Disable mode.

1) Disable Auto Join option on UA1.
2) UA1 and UA2 are in call.
3) UA3 dial “*74<UA1 number>” to initiate Join call.
4) UA1 will get call waiting tone.
5) At UA1 press FLASH, then UA1, UA2 and UA3 will get in conference call.

7.2.11 Call Pick Up

This procedure explains how to pick up the ringing call at any other UA.

User Agents: UA1-Board, UA2-Board and UA3-Board

1) Enable Call Pickup at UA1, UA2 on web page.
2) Make a call from UA3 to UA2.
3) While UA2 is ringing, go OFFHOOK on UA1 and dial “*67<UA2 Number>”, to pickup the call at UA1.
4) UA1 and UA3 will be in call i.e. UA1 has picked up the call ringing at UA2.

7.2.12 Call Back On Busy (CBOB)

This procedure explains how to call the busy extension without having to periodically try.

User Agents: UA1-Board, UA2-Board and UA3-Board

1) At UA1, enable call back on busy on webpage.

2) UA2 and UA3 are in call.

3) Make call from UA1 to UA2.

4) If UA2 doesn't answer the call,

5) Go on hook On UA1 then go off hook dial “*59” and hang up.

6) When UA2-UA3 call ends

7) UA1 rings. When you go OFFHOOK at UA1, UA2 is called automatically. That is UA2 rings, if UA2 accepts the call by Off Hook.

7.2.13 Attended Call Transfer

When there are two calls, this procedure explains how to transfer the call to between other 2 ends.

User Agents: UA1-Board, UA2-Board and UA3-Board

1) UA1 and UA2 are in call.

2) UA1 presses FLASH and dials UA3, UA3 accepts the call.

3) UA1 and UA3 are in call.

4) UA1 transfers the call by dial “*45”.

5) UA1 should get busy tone.

6) UA2 and UA3 should be in call.

7.2.14 Normal Call Transfer

This procedure explains how to transfer the existing call to a third UA.

User Agents: UA1-Board, UA2-Board and UA3-Board

1) UA1 and UA2 are in call.

2) UA1 transfers the call to UA3 by dialing “*49<UA3 number>”.

3) UA1 should get busy tone

4) UA3 starts ringing, UA3 accept the call.

5) Call should connect between UA2 and UA3.

7.2.15 Auto Attended Transfer

This is a special case of attended transfer where UA hangs up the phone which leads to the call being transferred between other 2 UAs.

User Agents: UA1-Board, UA2-Board and UA3-Board

1) On UA1 enable Auto attended transfer on web page,
2) UA1 and UA2 are in call.
3) UA1 presses FLASH and dial UA3.
4) UA1 and UA3 are in call.
5) UA1 putting UA2 and UA3 in conference by dial "*46". All three are in conference.
6) Hang up at UA1.
7) The call should connect between UA2 and UA3.

7.2.16 Call Forward On Busy (CFOB)

This procedure explains how to setup the call forward on busy condition and what happens when there is an Incoming call.

1) UA1 sets CFOB URI as UA2 (on web page).
2) Make sure UA1 is busy (i.e. call waiting is not possible at UA1).
3) Made call from UA3 to UA1.
4) UA1 forwards call to UA2, UA2 starts ringing.

7.2.17 Call Forward On No Answer (CFONA)

This procedure explains how to setup the call forward on no answer condition and what happens when there is an incoming call.

1) UA1 sets CFOB URI as UA2 (on web page).

2) UA3 calls UA1. UA1 does not answer.

3) The call is forwarded to UA2 after a configured no answer time out.

7.2.18 Call Forward Unconditional (CFU)

This procedure explains how to setup the unconditional call forward and what happens when there is an incoming call.

1) UA1 sets CFU URI as UA3

2) UA2 calls UA1

3) The call is forwarded to UA3 i.e. UA3 starts ringing

7.2.19 Call mute / Call un-mute

Call mute

1) UA1 and UA2 are in call.

2) On UA1 dial “*431”.

3) UA1 should hear UA2's voice. On UA2 silence.

Call un-mute

1) UA1 and UA2 are in call.

2) On UA1 dial “*431”.

3) Both ends should hear voice.

7.2.20 Do Not Disturb (DND)

DND is the condition when all incoming call will get busy indication. If call forward on busy condition is already setup, then the call will get diverted to that UA.

To enable

1) On UA1 dial “*441”.

2) Call from UA2 to UA1.

3) UA2 should get busy tone.

To disable

1) To Disable DND : On UA1 dial “*440”.

2) Call from UA2 to UA1.

3) UA1 should start ringing.

7.2.21 Outgoing Call Barring

This setting will disable making of outgoing calls from this FXS port.

To Enable

1) On UA1 dial “*481”.

2) Call from UA1 to UA2.

3) Call should not connect to UA2.

To Disable

1) On UA1 dial “*480”.

2) Call from UA1 to UA2.

3) UA2 should start ringing.

7.3 Supplementary Service

7.3.1 Anonymous Call Barring

This feature automatically rejects the incoming calls with anonymous caller id.

1) On UA1 enable Anonymous Call Barring on web page.
2) On UA2 disable Outgoing Caller ID option on web page.
3) Make call from UA2 to UA1.
4) UA1 rejects the call because it is call with Anonymous caller id.

7.3.2 Secretary Services

This feature ensures that only calls received from secretary or transferred by secretary are allowed.

1) Enable secretary service at UA1 and configure secretary URI as UA2.
2) Now make a call from UA3 to UA1.
3) Call connected between UA2 (diverted to secretary) and UA3.
4) UA2 can do Normal Transfer / Attended Transfer call to UA1.
5) Only secretary UA2 can call UA1 directly.

7.3.3 Absent Service

This feature will ensure that for all incoming calls a recorded message will be played. Typically, recorded message informs callers that the user is unavailable.

1) Enable Absent service on UA1.
2) Make an incoming call to UA1 from UA2.
3) On UA2 should hear recorded absent message.

7.3.4 Music On Hold

This feature ensures that pre-recorded music is played when remote end is put on hold.

1) Enable music on hold on UA1.
2) UA1 calls UA2, UA2 accept the call.
3) Press Flash on UA1.
4) UA2 should hear music.

7.3.5 Wakeup Alarm Service

This feature allows user to set up to 5 wakeup alarms along with its respective messages. When time reaches the set alarm time, analog phone connected to the FXS port starts ringing (if it is On Hook). And the caller id (if device is connected) will display the message specified for this alarm.

1) On UA1, set up wake up reminders by specifying date , time and alarm message
2) When UA1 time matches the alarm time, UA1 starts ringing.
3) Go off hook to hear wake up message.

NOTE: Alarm will **NOT** be heard if the handset is already Off Hook.

7.3.6 Group Dial

This feature allows groups of users to be created. Each group has an identifier number and set of URIs that are part of the group. Up to 5 member URIs can be specified for each group. Up to 50 group entries can be specified for each FXS port. Whenever dialed digit sequence matches the group number, call is made to each member URIs. Whichever member picks up the call first, is connected with the caller. Rest all calls are terminated.

1) Enable group dial at UA1.
2) Specify group dial no and enter up to five group members.
3) UA1 goes off hook and dial <group dial number>.

4) All the members should start ringing.

5) Accept the call at any one user, call gets connected and the other users should stop ringing.

7.3.7 Hot Link

This feature is similar to group dial except that there is no group number. When UA goes Off Hook, the call is made to each of the hot link member. Up to 10 hot link members can be specified. Two variants of hot link are supported – hotline or warmline. In hotline, the calls are made as soon as UA goes Off Hook where as in other case, the calls are made if no number is pressed by the user and inter digit timeout occurs. Only one of these hot link type can be set for each UA.

1) Enable hotline at UA1 and enter hot link user names / uris

2) UA1 goes off hook and all the users should start ringing.

3) Accept the call at any one user, that user gets connected and the other users stop ringing.

7.3.8 Abbreviated Dialing

This is a quick dial setting. Essentially abbreviated dial numbers can be set for various uris allowing user to dial only short number for frequently dialed URIs. It also enables alpha numeric SIP uris to be dialed via analog phone by specifying a short number for such SIP URI. Up to 100 abbreviated dial entries can be specified for each FXS port. This feature is also known as address book feature.

1) Enable abbreviated dialing at UA1.

2) There are two columns, first column for abbreviated number and second column for URI.

3) Dial one of these numbers and it should call the specified sip URI.

7.4 FAX Services

This section describes how to send and receive fax in pass through mode as well as T38 mode.

7.4.1 Pass-Through Fax

For sending and receiving fax in pass through mode (requires G711u/a codec), ensure that only G711u and G711a codecs are selected in the configuration. This is to ensure that G711u/a codecs are available for initiating fax pass through.

7.4.1.1 Sending Fax

User Agents: UA1 - Board, UA2 – Any SIP ATA (with fax machines connected to each ATA).

1) Assuming that UA1 and UA2 are registered to Proxy.
2) Dial the remote number on fax machine (optionally followed by #).

   e.g. 888# or dial 888 and wait for inter-digit timeout.
3) UA2's fax machine indicates the incoming fax call.
4) Now fax pages should get transmitted from originator fax machine to receiver of the call.

7.4.1.2 Receiving Fax

Same procedure as above, where UA2 is receiving the fax.

7.4.2 T38 Fax

For sending and receiving fax using T38, make sure that use of T38 is enabled in the configuration of both ATA end points involved in fax transmission.

7.4.2.1 Sending Fax

This procedure is exactly identical to that of sending fax in pass through mode.

7.4.2.2 Receiving Fax

This procedure is exactly identical to that of receiving fax in pass through mode.

7.5 *Features Supported via Keypad*

This section summarizes the key pad operations possible.

These are divided in two parts

1. Making configuration changes
2. Exercising call features

This section describes the configuration settings that can be changed from the keypad along with the procedure to make these changes. The configuration change is possible only after user pressing the special sequence to enter in configuration change mode. This special mode is excited by pressing another complementary sequence of keys or by user hanging up the phone.

7.5.1 Activate/Deactivate Configuration via keypad

User needs to press this sequence in order to enter/exit into special configuration change mode. Configuration via keypad also deactivated when receiver is OnHook

Syntax: *99<0/1>

Where,

0 - Deactivate configuration via keypad

1 - Activate configuration via keypad

Example:

*991 Deactivate configuration via keypad

*990 Activate configuration via keypad

7.5.2 Enable/Disable CFOB (Call Forward On Busy)

Syntax: *12<0/1>

Where,

0 - Disable CFOB

1 - Enable CFOB

Example:

*121 enable CFOB

*120 disable CFOB

7.5.3 Setting CFOB URI (Call Forward On Busy URI)

Syntax: *13*<cfob_uri>

Where,

cfob_uri - Uri where call to be forwarded when CFOB feature is enabled.

Example:

*13*11100, where 11100 is CFOB number

7.5.4 Enable/Disable CFU (Call Forward Unconditional)

Syntax: *14<0/1>

Where,

0 - Disable CFU

1 - Enable CFU

Example:

*141 enable CFU

*140 disable CFU

7.5.5 Setting CFU URI

Syntax: *15*<cfu_uri>

Where,

cfu_uri - URI where call to be forwarded when CFU feature is enabled.

Example:

*15*11100, where 11100 is CFU number

7.5.6 Enable/Disable CFONA (Call Forward On No Answer)

Syntax: *16<0/1>

Where,

0 - Disable CFONA

1 - Enable CFONA

Example:

*161 enable CFONA

*160 disable CFONA

7.5.7 Setting CFONA URI

Syntax: *17*<cfona_uri>

Where,

cfona_uri - Uri where call to be forwarded when CFONA feature is enabled.

Example:

*17*11100, where 11100 is CFONA number

7.5.8 Setting CFONA Timer

Syntax: *18*<time_insec>

Where,

time_insec - Time (in seconds) when elapsed, call is to be forwarded when CFONA feature is enabled.

Example:

*18*30, where 30 is CFONA time

7.5.9 Enable/Disable Auto Attendant Transfer

Syntax: *19<0/1>

Where,

0 - Disable auto attendant transfer

1 - Enable auto attendant transfer

Example:

*191 enable auto attendant transfer

*190 disable auto attendant transfer

7.5.10 Enable/Disable Anonymous Call Barring

Syntax: *20<0/1>

Where,

0 - Disable anonymous call barring

1 - Enable anonymous call barring

Example:

*201 enable anonymous call barring

*200 disable anonymous call barring

7.5.11 Enable/Disable Outgoing Caller ID

Syntax: *21<0/1>

Where,

0 - Disable outgoing caller ID

1 - Enable outgoing caller ID

Example:

*211 enable outgoing caller ID

*210 disable outgoing caller ID

7.5.12 Enable/Disable Call Pickup

Syntax: *22<0/1>

Where,

0 - Disable call pickup

1 - Enable call pickup

Example:

*221 enable call pickup

*220 disable call pickup

7.5.13 Enable/Disable Call Waiting

Syntax: *23<0/1>

Where,

0 - Disable call waiting

1 - Enable call waiting

Example:

*231 enable call waiting

*230 disable call waiting

8 Using ATA Features: FXO

In addition to traditional ATAs that support FXS lines, one of Micrel's evaluation boards (ATA-FXSO) also includes an FXO port to support failover. One of the ways the FXO port can be used is in case the VoIP endpoint functions are not available on the FXS port – for example, the network is down, or the SIP server does not respond.

IMPORTANT: Please read Section 7 before reading this section – since this section provides details of how the ATA-FXSO configuration differs from the FXS-only systems and assumes you have read how to configure FXS-only systems.

For the ATA-FXSO board there are three additional items to understand:

1) If you use 9 as the dial prefix, you get access to your PBX line (where the FXO is connected to). This way you can use your Phone connected to the FXS port simply as a regular analog phone; it will support all the features that your PBX supports, including FAX, call waiting, etc.
2) Another important aspect is failover – the FXS phone is configured for failover as follows:
   a) If you use SIP server 1 (using prefix "1") and the server does not respond, the system fails over to the FXO port and connects to the number dialed via the FXO line.
   b) Same as above for SIP server 2 (using prefix "2").
3) There are three additional configuration items provided
   a) Connect incoming FXO call directly to local FXS. If enabled, incoming calls on the FXO port are routed to the FXS line, similar to a regular analog phone.
   b) Play voice message when there is incoming call on FXO line (this is applicable when configuration item described in above is disabled). This is enable or disable flag. If enabled, it will play message otherwise dial tone will be played out, to indicate that caller needs to dial VoIP number.
   c) Play voice message when there is incoming VoIP call on FXO port. If enabled, it will play message otherwise dial tone will be played out, to indicate that caller needs to dial PSTN / PBX number.
4)

8.1 Outbound Call from FXS Port on SIP Server 1

User Agents: UA1 (FXS) - Board, UA2 – Any SIP Phone

1) Go Off-hook on UA1 and dial UA2 but by forcing call through SIP Server 1, i.e., dial the prefix associated with Server 1 first and then the rest of the number.
2) Ringing tone should be heard in UA1 and UA2 should ring.
3) Go Off-hook on UA2 after one ring and verify Caller ID (UA1's phone number) is displayed on UA2.
4) Verify call is established between UA1 and UA2.
5) Go On-hook on UA2 to end call.
6) A busy tone should be heard on UA1.
7) Go On-hook on UA1.

8.2 Outbound Call from FXS Port on SIP Server 2

User Agents: UA1 - Board, UA2 – Any SIP Phone (registered on SIP server 2, along with UA1)

1) Go Off-hook on UA1 and dial UA2 but by forcing call through SIP Server 2, i.e., dial the prefix associated with Server 2 first and then the rest of the number.
2) Ringing tone should be heard in UA1 and UA2 should ring.
3) Go Off-hook on UA2 after one ring and verify Caller ID (UA1's phone number) is displayed on UA2.
4) Verify call is established between UA1 and UA2.
5) Go On-hook on UA2 to end call.
6) A busy tone should be heard on UA1.
7) Go On-hook on UA1.

8.3 Inbound Call on FXS Port via SIP Server 1

User Agents: UA1 - Board, UA2 – Any SIP Phone

1) Go Off-hook on UA2 and dial UA1.

2) Ringing tone should be heard in UA2 and UA1 should ring.

3) Go Off-hook on UA1 after one ring and verify Caller ID (UA2’s phone number) is displayed on UA2.

4) Verify call is established between UA1 and UA2.

5) Go On-hook on UA1 to end call.

6) A busy tone should be heard on UA2.

7) Go On-hook on UA2.

8.4 Inbound Call on FXS Port via SIP Server 2

User Agents: UA1 - Board, UA2 – Any SIP Phone (registered on SIP server 2, along with UA1)

1) Go Off-hook on UA2 and dial UA1.

2) Ringing tone should be heard in UA2 and UA1 should ring.

3) Go Off-hook on UA1 after one ring and verify Caller ID (UA2’s phone number) is displayed.

4) Verify call is established between UA1 and UA2.

5) Go On-hook on UA1 to end call.

6) A busy tone should be heard on UA2.

7) Go On-hook on UA2.

8.5 Outbound call from FXO Port

In this step, we will dial out directly through the FXO port; to do so, first make a note of the FXO Dial Prefix. In the following, we assume that you have set up the FXO Dial Prefix to be “9”. (Note: The FXO Dial Prefix is configurable through the WebGUI, as explained in the Configuration User’s Manual).

User Agents: UA1 (FXS) - Board, UA2 – Any landline phone

1) Go Off-hook on UA1 and dial 9-X-YYYYYYY# where X is the PBX switching system Prefix Number, and YYYYYYYY is any external landline or cellphone (UA2).

2) Ringing tone should be heard in UA1 and UA2 should ring.

3) Go Off-hook on UA2 after one ring and verify Caller ID (PBX Name and Phone Number) is displayed on UA2.

4) Verify call is established between UA1 and Ua2.

5) Go On-hook on UA2 to end call.

6) UA1 should behave like a regular landline – if you do not hang up you will get a message “If you want to make a call….”. If you still do not hang up you will get a fast busy.

7) Go On-hook on UA1.

8.6 Inbound call on FXO Port (Connect to local FXS)

In this step we will dial directly to the ATA FXO port. To do this demo, you will need to know the number that has been assigned to the FXO Port by the PBX Switching system.

User Agents: UA1 (FXS) - Board, UA2 – Any landline phone

1) Go Off-hook on UA2 and dial the phone number that has been assigned to the FXO port of the ATA.

2) Ringing tone should be heard in UA2 and UA1 should ring.

3) Go Off-hook on UA1.

4) Verify call is established between UA1 and UA2.

5) Go On-hook on UA2 to end call.

6) Go On-hook on UA1.

8.7 Inbound call on FXO Port (Connect to any SIP phone, with message)

In this step we will connect to VoIP endpoint from PBX extension or external PSTN number. FXO port has its own configuration in VoIP world where it is assigned unique userid (similar to FXS port). To do this demo, you will need to know the number that has been assigned to the FXO Port by the PBX Switching system. Make sure you have disabled connect to local fxs and enabled playing message for incoming PSTN/PBX call.

User Agents: UA1 (FXS) – Any SIP phone, UA2 – Any landline phone

1) Go Off-hook on UA2 and dial the phone number that has been assigned to the FXO port of the ATA.
2) Message should be heard in UA2 asking caller to dial VoIP number.
3) UA2 dials VoIP number of UA1
4) Go Off-hook on UA1.
5) Verify call is established between UA1 and UA2.
6) Go On-hook on UA1 and/or UA2 to end call.
7) Go On-hook on UA1.

8.8 Inbound call on FXO Port (Connect to any SIP phone, without message)

In this step we will connect to VoIP endpoint from PBX extension or external PSTN number. FXO port has its own configuration in VoIP world where it is assigned unique number (similar to FXS port). To do this demo, you will need to know the number that has been assigned to the FXO Port by the PBX Switching system. Make sure you have disabled connect to local fxs and enabled playing message for incoming PSTN/PBX call.

User Agents: UA1 – Any SIP phone, UA2 – Any landline phone

1) Go Off-hook on UA2 and dial the phone number that has been assigned to the FXO port of the ATA.
2) Dial tone should be heard in UA2 indicating caller to dial VoIP number.
3) UA2 dials VoIP number of UA1
4) Go Off-hook on UA1.
5) Verify call is established between UA1 and UA2.
6) Go On-hook on UA1 and/or UA2 to end call.
7) Go On-hook on UA1.

8.9 Outbound call on FXO Port (from any SIP phone, with message)

In this step we will connect from any VoIP endpoint to PBX extension or external PSTN number. FXO port has its own configuration in VoIP world where it is assigned unique number (similar to FXS port). To do this demo, you will need to know the number that has been assigned to the FXO Port by the VoIP system. Make sure you have enabled playing message for incoming VoIP call.

User Agents: UA1 – Any SIP phone, UA2 – Any landline phone

1) Go Off-hook on UA1 and dial the VoIP phone number that has been assigned to the FXO port of the ATA.
2) Message should be heard in UA1 telling caller to dial PBX/PSTN number.
3) UA2 dials the number of UA1 (and it should start ringing)
4) Go Off-hook on UA1.
5) Verify call is established between UA1 and UA2.
6) Go On-hook on UA1 and/or UA2 to end call.
7) Go On-hook on UA1.

8.10 Outbound call on FXO Port (from any SIP phone, without message)

In this step we will connect from any VoIP endpoint to PBX extension or external PSTN number. FXO port has its own configuration in VoIP world where it is assigned unique number (similar to FXS port). To do this demo, you will need to know the number that has been assigned to the FXO Port by the VoIP system. Make sure you have enabled playing message for incoming VoIP call.

User Agents: UA1 – Any SIP phone, UA2 – Any landline phone

1) Go Off-hook on UA1 and dial the VoIP phone number that has been assigned to the FXO port of the ATA.
2) PBX/PSTN dial tone should be heard in UA1 indicating caller to dial PBX/PSTN number.
3) UA2 dials the number of UA1 (and it should start ringing)
4) Go Off-hook on UA1.

5) Verify call is established between UA1 and UA2.

6) Go On-hook on UA1 and/or UA2 to end call.

7) Go On-hook on UA1.

8.11 Redirection of Outbound Call from VoIP to FXO

This step shows the capability of the ATA-FXSO to switch from the FXS port to the FXO port in case there is no connectivity to the SIP Servers.

User Agents: UA1 - Board, UA2 – Any landline phone

1) Unplug Ethernet cable that is connected to the ATA-FXSO and wait about 1 minute (to detect connection change/reconfiguration)

2) Go Off-hook on UA1.

3) Dial 1-X-YYYYYYY# where 1 is the FXS Server1 Prefix, X is the PBX switching system Prefix Number, and YYYYYYYY is any external landline or cellphone (UA2). By dialing this number you are forcing to use the FXS port of the ATA-FXSO, but since the FXS port cannot contact the SIP Server for making call, the ATA-FXSO will switch over to the FXO port to complete the call.

4) Ringing tone should be heard in UA1 and UA2 should ring.

5) Go Off-hook on UA2 after one ring and verify Caller ID (PBX Name and Phone Number) is displayed on UA2.

6) Verify call is established between UA1 and UA2.

7) Go On-hook on UA2 to end call.

8) UA1 should behave like a regular landline – if you do not hang up you will get a message “If you want to make a call….”. If you still do not hang up you will get a fast busy.

9) Go On-hook on UA1.

9 ATA Universal Board Description

This section provides a detailed overview of functional areas of the board. Use this section as a reference for getting familiar with features and functionality.

9.1 Evaluation Board Features

The KSZ8342Q Evaluation Board contains the following features:

- IP Phone with Ethernet, headset, handset, and speaker interfaces
- Two 10/100 Ethernet ports
- 32 MB of SDRAM memory
- Eight 8 MB of SPI NOR Flash memory
- Communications port utilizing a UART and one DB-9 connector
- Access to GPIO pins and LED indicators
- One JTAG Port for the MIPS processor
- Interface to an external SLIC daughter board
- 128-pin QFP KSZ8342Q device

9.2 Top Level Evaluation Board Organization

The block diagram in Figure 2 shows the functionality and sub areas on the KSZ8342Q Universal ATA evaluation board. Every signal is not shown on this block diagram. It is advised that the developer look at the schematic of the board to get familiarized with the items shown on the block diagram and the detailed items on the board. The remainder of the sections which follow in Section 7 will provide the necessary details of each feature.

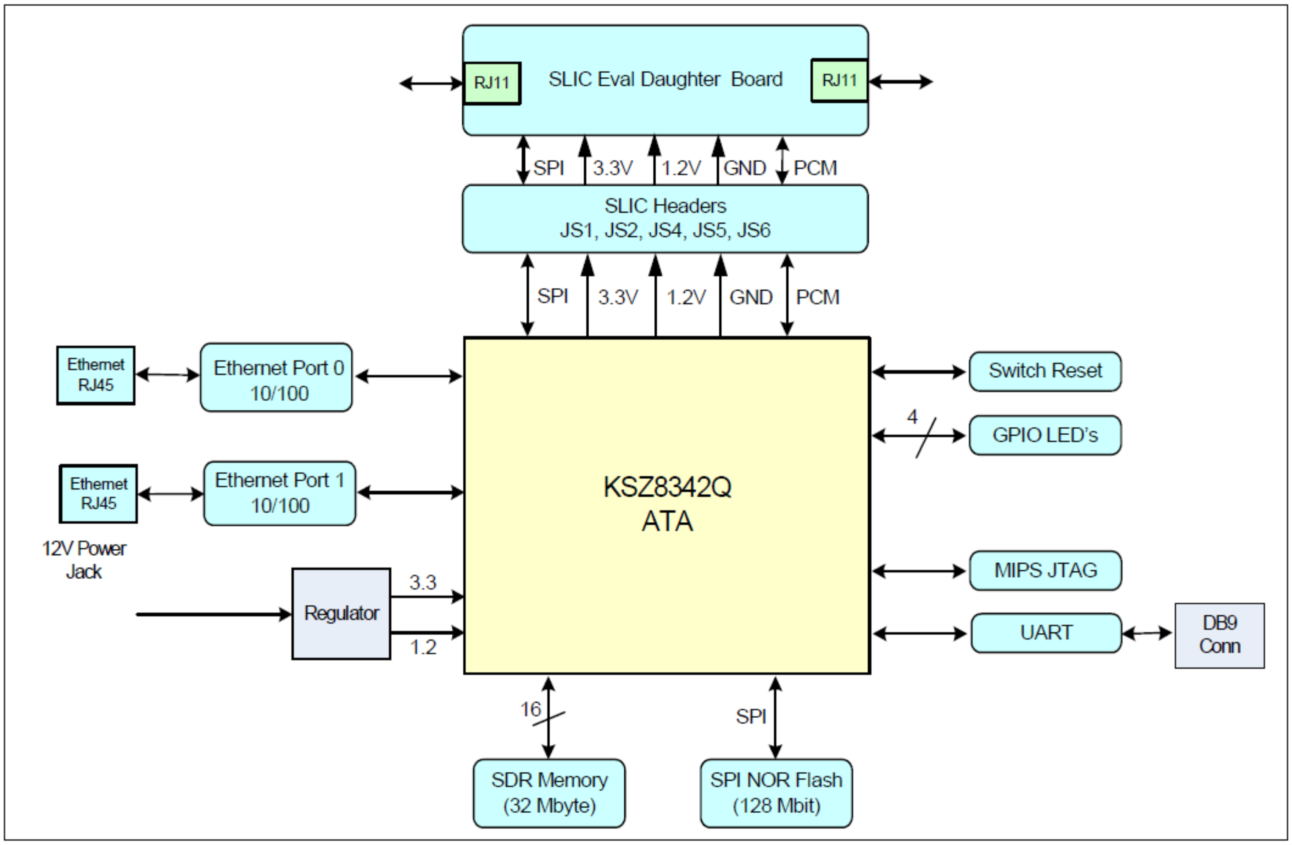


Figure 2: Block Diagram of the KSZ8342Q VOIP Evaluation Board

Figure 3 illustrates the actual board with the component side showing.

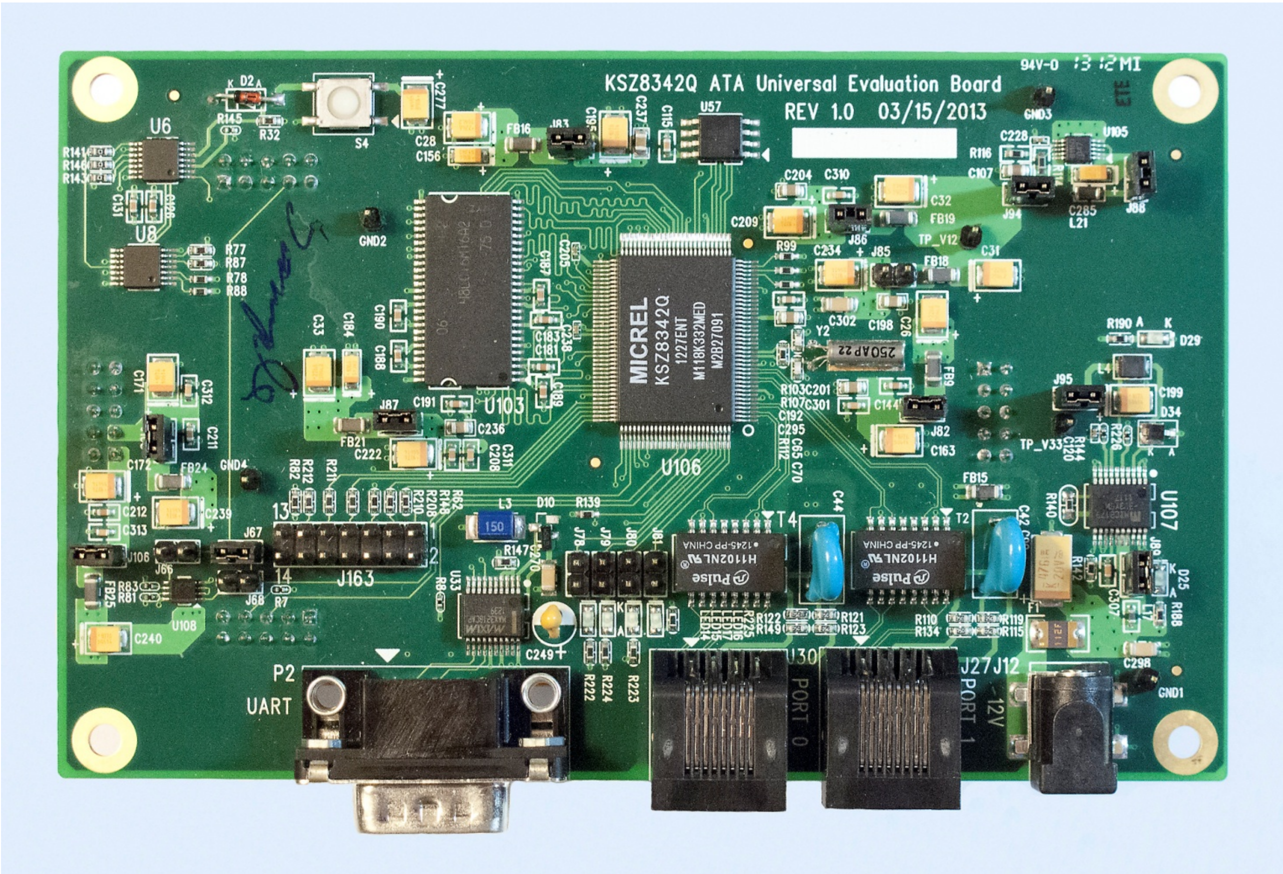


Figure 3: Layout View of the KSZ8342Q Universal ATA Evaluation Board

9.3 Memory

The KSZ8342Q device supports two types of memory; single data rate SDRAM and serial NOR Flash memory. The sections which follow provide detailed information on its usage.

9.3.1 SDRAM Memory

The SDRAM memory is used by the MIPs processor to execute the code. At power up time, the code image that is in the Serial NOR Flash device is loaded into the SDRAM for execution by the MIPs processor. When power is turned off, the contents of the SDRAM are lost.

- The memory data width required is 16 bits. At this time, only 16-bit devices are supported.

- The minimum - maximum SDRAM size that can be used is 8 Mbytes - 64 Mbytes. The Eval board has 32 Mbytes installed.
- The SDRAM is located at U103.

Refer to the latest firmware readme files and SDK package for more details on the memory organization and utilization.

9.3.2 Serial NOR Flash Memory

The serial NOR Flash memory is used to store code image. It is programmed at the factory with the latest revision of code. At power up time, the code image is loaded into the SDRAM for execution.

- The maximum Serial NOR Flash Memory size that can be used is 16 Mbytes.
- The Serial NOR Flash memory uses the SPI interface.

The KSZ8342Q universal ATA evaluation board provides a Macronix 128 Mbit serial NOR Flash memory (MX25L12836E). This device is mounted directly onto the board at location U57. The evaluation board is configured to boot from NOR flash.

Refer to the latest firmware readme files and SDK package for more details on the memory organization and utilization.

9.4 Ethernet Interface

The KSZ8342Q VoIP evaluation board supports two 10/100 Ethernet ports. These ports are available on two RJ45 connectors located at J27 and J30 along the bottom edge of the board. The internal Ethernet switch design is based on Ethernet switch technology available on standalone Micrel switch products such as the KSZ8773. The PHY design is based on Micrel’s KSZ8051/KSZ8091 standalone designs. Refer to the KSZ8342Q datasheet for more details on the available Ethernet functionality. It is recommended to use at least a Cat5e cable or better for all Ethernet connections.

9.5 PCM and SPI ATA Interfaces

The KSZ8342Q device incorporates the PCM interface to communicate with the SLIC module. The signals include PCLK, PFSYNC, PCMCSN, RXD, TXD, SPI SCLK, SPI SDI, SPI SDO, and SLIC Reset. These signals are connected to the connector pins which mate with the corresponding connectors on the SLIC Eval Daughter Board. See Table 4 for the details of the specific locations of these signals.

9.6 Communications Interface

Communication with the onboard processor and firmware takes places across a serial interface communication port. An onboard UART (Maxim MAX3218) handles the communication across the RS232 serial bus. A male DB9, null modem connector is located at location P2 at the lower-left side of the board. The nominal settings for communication are 8 bits, no parity, and 1 stop bit (8N1) at 115200 baud.

This is the interface that the user will use to communicate to the processor and onboard firmware from an external computer. The firmware update process can also utilize this interface. When the board is first manufactured, the first code image will be transferred to the onboard Flash device via the JTAG interface.

9.7 MIPS JTAG

The KSZ8342Q VoIP evaluation board supports JTAG operations for the MIPS CPU, The MIPS JTAG interface is a 14-pin header located at header J163. The JTAG interface can be used to download code to the NOR Flash memory as well as for software development and debugging purposes. The header is located near the RS232 connector.

9.8 GPIO Pins

The KSZ8342Q universal ATA evaluation board supports four of the seventeen GPIO pins available on the KSZ8342Q device. SGPIO[22:19] are brought out to LED indicators LED[17:14]. A logic low of an individual GPIO pin turns the associated LED ON and a logic high turns it OFF. These LED's can be used to assist with software debug and development. The LEDs can be removed if desired and the GPIO pin used for any general I/O purpose as necessary.

Table 2 shows the relationship between the GPIO pins and the LEDs on the board.

Table 2: GPIO Pin to LED Mapping

SGPIO	LED	Jumper	Relative Location on Board
19	14	J78	LEDs and jumpers located to the right of the DB9 connector near the bottom of the board.
20	15	J79	
21	17	J80	
22	16	J81	

9.9 Reset

There are three types of resets implemented on this evaluation board.

- Power on reset
- Push button reset
- Watchdog Timer reset

9.9.1 Power On Reset

A power on reset circuit is implemented which maintains a zero voltage level on the Reset pin (Pin 32) of the KSZ8342Q device for approximately 100ms while the various voltage regulators and other devices are powering up.

9.9.2 Push Button Reset

A momentary push button switch (S4) is provided to enable the user to power cycle the KSZ8342Q device at any time. S4 is located at the top-left edge of the board.

9.9.3 Watchdog Timer Reset

The board has been designed to make available the optional use of an internal Watchdog timer to assert reset after an elapsed programmed period of time. When a specific internal programmable timer expires, the WRSTO signal output on Pin 103 is used to create a reset signal to the KSZ8342Q device. Table 3 presents the options the user has when using this Watchdog timer reset function. It is recommended that the jumpers remain in the default state until thoroughly understood.

Table 3: Watchdog Timer Jumper Settings

Jumper	Jumper Setting (Default)	Description
J90	Open	This jumper is used to match the polarity of the watchdog timer reset output. If the KSZ8342Q device is programmed with an active high watchdog timer reset output, this jumper should be installed. The default setting is an active low WRSTO reset. In this case a high voltage on WRSTO is used to reset the board.
J91	Closed	This jumper is used to match the polarity of the watchdog timer reset output. If the KSZ8342Q device is programmed with an active low watchdog timer reset output, this jumper is installed. The default setting is an active low WRSTO reset. In this case a low voltage on WRSTO is used to reset the board.
J93	Open	Installed = The watchdog timer reset is not used to reset the board. Not installed = The watchdog timer reset is used to reset the board.

9.10 SLIC Evaluation Daughter Board Interface

The Micrel KSZ8342Q universal ATA evaluation board interfaces to an external SLIC evaluation daughter board through a set of five inline connectors (JS1, JS2, JS4, JS5, and JS6). The mating portions of each of the connectors are located on the bottom of each of the two evaluation boards. The connectors are oriented in a specific manner to make lining up the boards easy. When the two boards are connected together, the component side of each board is accessible for user interaction. Figure 4 shows the two boards connected together.

:

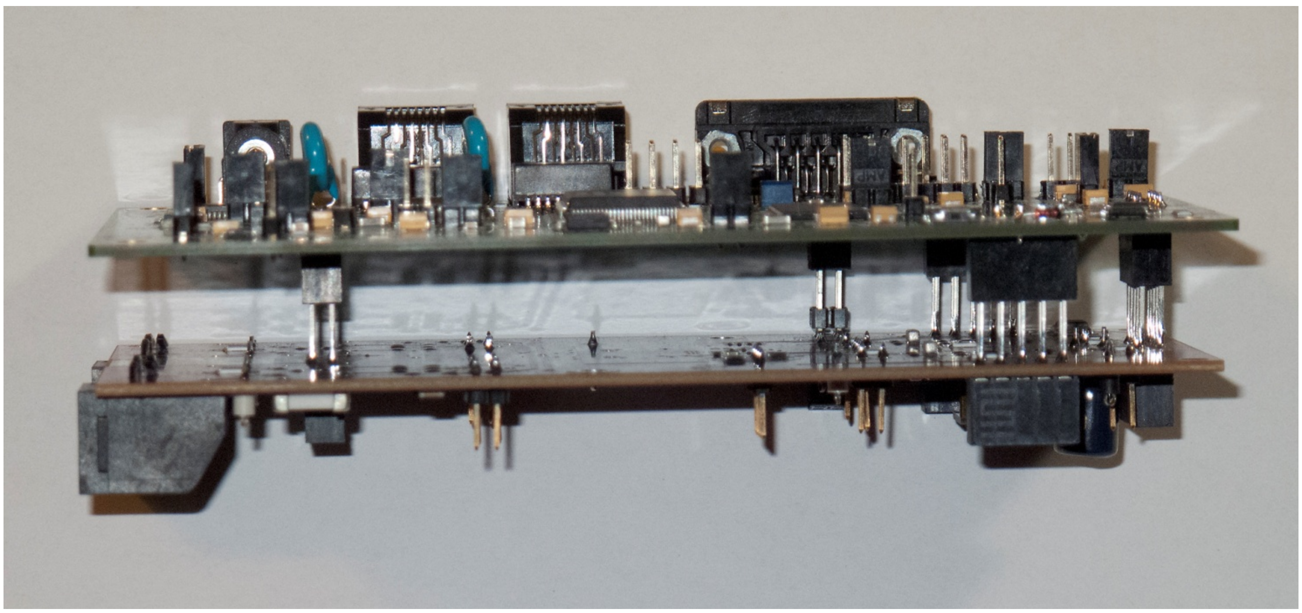


Figure 4: Connecting the KSZ8342Q and SLIC Evaluation Daughter Boards Together

The functionality of each of the connector pins is Table 4.

Table 4: SLIC Daughter Board Interface Connector Pinout

Micrel Connector	Pin Number	Functionality
	1	N/C

Micrel Connector	Pin Number	Functionality
JS1	2	GND
	3	LCDD0/SPIDI
	4	GND
	5	LCDD2/SP3CK
	6	GND
	7	LCDD2/SPISPN
	8	GND
	9	/INT GPIO PIN LCDD3
	10	GND
JS2	1	LCDEN/SPIDO
	2	GND
	3	1.2 VDC
	4	GND
JS4	1	1.2 VDC
	2	GND
	3	3.3V VCC
	4	GND
	5	N/C
	6	GND
	7	N/C
	8	GND
	9	N/C
	10	GND

Micrel Connector	Pin Number	Functionality
JS5	1	N/C
	2	GND
	3	GND
	4	GND
	5	N/C
	6	GND
	7	GND
	8	GND
	9	N/C
	10	GND
	1	KCOL0/PCLK
JS6	2	GND
	3	KCOL3/PTXD
	4	GND
	5	KOL2/PRXD
	6	GND
	7	FSYNC SLIC
	8	GND
	9	KCOL5
	10	GND

9.11 Narrow Band and Wideband Selection

The board can be configured for narrow band or wideband operation. There are two jumpers to configure the board for narrow band or wideband operation. The standard configuration is narrow band mode. Please contact Micrel for wideband usage. The Table below summarizes the configuration of the two jumpers.

Table 5: Narrow Band / Wide Band Selection Jumpers

Configuration	Jumper		
	J66	**J67**	**J68**
Narrow Band	Out	In	Out
Wideband	In	Out	In

9.12 *Power*

This section provides the information necessary to understand the voltages necessary to power the evaluation board, voltage configuration choices available, and voltage / power monitoring capabilities.

9.12.1 Primary Power Source

The KSZ8342Q Universal ATA evaluation board is powered by an external 12V power module. The board features a power LED indicator at location D29 (right side of board) so that the user may easily determine that 12V power is applied to the board and the 3.3V V33 is being generated. An external 12V power supply is required to be connected to J12. J12 is located to the right of the two Ethernet connectors. The user must select a configuration from one of the columns and make sure the jumpers are all correctly configured.

Table 6: Selecting the Power Source

Jumper	Power Jack PWR
J102	Open
J103	Open
J104	Closed

9.12.2 Voltages and Test Points

The 12V DC power to the evaluation board is converted down to two other voltages; 3.3V and 1.2V. The following tables outline the onboard derived voltages and associate test points.

Table 7: KSZ8342Q VoIP Evaluation Board Derived Voltages and Test Points

Voltages	Test Points	Location
V1.2 (Digital and analog core)	TP_V12	Located slightly below the board silkscreen title
V3.3 (Digital and analog core/I/O, and CODEC)	TP_V33	Located about one inch above the power input connector.
VDC 1.2V for the SLIC Eval Board		Connector JS4, pin 1. See Table 4 for more information.
VCC 3.3V for the SLIC Eval Board		Connector JS4, pin 3. See Table 4 for more information.
0V (GND)		GND1, GND2, GND3, GND4 (Various locations).

9.12.3 Power Related Jumpers and Test Points

The board has an assortment of jumpers which can serve as breakpoints and measurement points for the various related power sources and connections. These can be useful when measuring currents associated with specific parts of the device or board. These jumpers are listed in Table 8.

Table 8: Power Related Jumper Settings

Jumper Numbers	Description	Default Jumper Setting
J82	Use filtered V33 as source for VDDA33	Connected
J83	3.3V SDRAM power connection	Connected
J85	1.2V VDDA12 Analog power connection	Open
J86	1.2V VDDCORE connection	Connected
J87	3.3V VDDIO33 connection	Connected
J88	3.3V LDO input connection for power regulator	Connected

Jumper Numbers	Description	Default Jumper Setting
J89	12V LDO input connection for power regulator	Connected
J94	3.3V -> 1.2V power regulator output connection	Connected
J95	12V -> 3.3V power regulator output connection	Connected
J105	3.3V SLIC power connect	Connected
J106	1.2V SLIC power connect	Connected

9.13 Numerical List of Jumpers

The information in the following tables is provided as a convenient way of finding out what a specific jumper is used for and its default setting. A complete numerical listing of 'J' jumpers J1 - J163 for the KSZ8342Q universal ATA evaluation board is shown in Table 9. A complete numerical listing of 'JP' jumpers JP1 - JP111 is shown in Table 10. A complete numerical listing of 'P' test points P1 - P4 is shown in Table 11. In these tables, and NU designation indicates the jumper is not used.

Table 9: Numerical List of "J" Jumpers J1 – J163

"J" Jumper Number	Description	Default Jumper Setting
J1 – J2	N/U	-
J3	RJ11 telephone (SLIC Channel B) connector	Mounted
J4	RJ11 telephone (SLIC Channel A) connector	Mounted
J5 – J11	N/U	-
J12	12V Power Jack	Mounted
J13	MIC Logic Analyzer Connector	Mounted

"J" Jumper Number	Description	Default Jumper Setting
J14 – J26	N/U	-
J27	RJ45 – Ethernet Connector	Mounted
J28 – J29	N/U	-
J30	RJ45 – Ethernet Connector	Mounted
J31 – J52	N/U	-
J53	SLIC device current consumption - Insertion point for an ammeter to measure the current used by the SLIC device.	In
J54 – J65	N/U	See Table 4
J66	Wideband Select	
J67	Narrowband Select	
J68	Wideband Select	out
J69 – J77	N/U	-
J78	SGPIO[19] / LED14	out
J79	SGPIO[20] / LED15	out
J80	SGPIO[21] / LED17	out
J81	SGPIO[22] / LED16	out
J82	Use filtered Vss as source for VDDA33	in
J83	Use filtered V33 as source for VSDR33	in
J84	N/U	-
J85	Use filtered V12 as source for VDDA12	in
J86	Use filtered V12 as source for VDDCORE12	in
J87	Use filtered V33 as source for VDDIO33	in
J88	Select 5V input to supply 3.3V LDO Regulator	in

"J" Jumper Number	Description	Default Jumper Setting
J89	Select 5Vinpuit to supply 1.2V LDO Regulator	in
J90 – J104	N/U	-
J105	Use filtered V33 as source for VSLIC33	in
J106 – J162	N/U	-
J163	MIPS JTAG Interface – 14 Pin Header	Installed

Table 10 JP Jumper Information

"JP' Jumper Number	Description	Default Setting
JP1 – JP111	N/U	-

Table 11: P Connector Information

'P" Connectors	Description	Default Setting
P1	N/U	-
P2	Null-Modem Connector, DB9 Male Connector	Installed
P3	N/U	-
P4	N/U	-

10 SLIC Daughter Board Description

The information in this appendix provides information on at least some of the SLIC Daughter Boards available for use with the KSZ8342Q Universal ATA Evaluation Board.

10.1 SLIC Daughter Board – 2FXS

Vendor Part Number	Silicon Labs – Si32260-3217-PBI_20SLO-EVB Rev 2.0 Si3260PB20SL2-EVB Rev 1.0
SLIC Device Used	Silicon Labs – Si32260
Specific Revision Required	As above.
Number of telephone ports	2
Type of Port Support	FXS
Required Jumpers	None
Installation Instructions	No special installation steps are required as long as the SLIC daughter board is the correct revision, the SLIC device is the correct revision, and the onboard regulator scheme matches what is programmed into the Micrel firmware. Micrel incorporates a specific revision of Silicon Labs driver code into its firmware which matches a specific revision of SLIC daughter board. The board revision must be the one specified above. Contact Micrel regarding any attempt to run with a revision different than above.

10.2 SLIC Daughter Board – 1FXS

Vendor Part Number	Silicon Labs – Si3217xCFB10SLO-EVB Rev 1.0
SLIC Device Used	Silicon Labs – Si32171
Specific Revision Required	Assembly 1.2 -
Number of telephone ports	1
Type of Port Support	FXS
Required Jumpers	JP1, JP2 installed
Installation Instructions	No special installation steps are required as long as the SLIC daughter board is the correct revision, the SLIC device is the correct revision, and the onboard regulator scheme matches what is programmed into the Micrel firmware. Micrel incorporates a specific revision of Silicon Labs driver code into its firmware which matches a specific revision of SLIC daughter board. The board revision must be the one specified above. Contact Micrel regarding any attempt to run with a revision different than above.

10.3 SLIC Daughter Board – 1FXS+1FXO

Vendor Part Number	Silicon Labs – Si3217xFB-EVB Rev 2.1
SLIC Device Used	Silicon Labs – Si32178
Specific Revision Required	Rev 2.2
Number of telephone ports	2
Type of Port Support	1- FXS, 1-FXO

Required Jumpers	J21 1-2,
Installation Instructions	No special installation steps are required as long as the SLIC daughter board is the correct revision, the SLIC device is the correct revision, and the onboard regulator scheme matches what is programmed into the Micrel firmware. Micrel incorporates a specific revision of Silicon Labs driver code into its firmware which matches a specific revision of SLIC daughter board. The board revision must be the one specified above. Contact Micrel regarding any attempt to run with a revision different than above.

11 Firmware: Build and Update

The evaluation board comes with firmware pre-installed and is ready to use. In case an update is needed, this section describes how to build the ATA Evaluation System firmware and update the board with this firmware.

The firmware consists of three main parts:

1) Boot loader
2) Kernel
3) Application

11.1 Build the Firmware

11.1.1 Access the Source Tree and Documentation

The firmware and documentation is available via Github (https://github.com/). To access the firmware and the documentation:

- Have your Github account ready. In case you do not have one, create a Github account as follows:
    - click on the Github link above
    - select user name and password
    - follow instructions from Github
- Once you have an account, request Micrel to provide you with access – email requests to Micrel Support (voip-support@micrel.com).
- For an organization, Micrel can create a team access. In case you have multiple users, please provide the Github user names for all persons requesting access.

11.1.2 Build Process

Once you have access to the source tree in Github:

- Clone the source tree on your development PC

- Review the instructions at the top-level directory
    - Readme: describes the steps need to build
    - Makefile: describes the build process in detail
- With a successful build, you will have a new image which can be downloaded to the ATA Evaluation board. Follow the steps described in the appropriate sections below to download this image.

11.2 Update the Firmware

11.2.1 Setup

Same general environment as described in Figure 1. Any telephones or extra evaluation boards do not have to be installed. The important part is to make sure the computer, the Ethernet connection, and the RS232 connection between the computer and the board are connected and functioning.

- New Firmware Image file, located in a directory of your choice.
- SolarWindsTrivial FTP Program.
- Communication Utility (Hyperlinks, TeraTerm, or equivalent).
- The SolarWinds TFTP server can be downloaded from the following web address.

  http://www.solarwinds.com/products/freetools/free_tftp_server.aspx

11.2.2 Update Kernel and Application

The procedure described below outlines the steps that are needed to perform the Kernel+Application software upgrade.

11.2.2.1 Image

An image has the following characteristics

1) Name -- INAME

2) Date: IDATE

3) Filename - FNAME

11.2.2.2 Get Current Configuration Information

In this step, you will setup the environment to burn the new image to flash memory.

1) Note ATA Settings

   Prior to the upgrade you need to record the previous SIP Server and extensions for the ATA-2FXS board. You can record it by following the steps below:

   a) Get the IP Address of ATA-2FXS

      i) Connect the PC's serial port to the ATA-2FXS using the null modem serial cable

      ii) Open the Terminal emulator (with 115200, 8N1-None settings) – so that you will be able to see the ATA-2FXS boot up messages.

      iii) Power up the ATA-2FXS – you will see messages in the terminal window

      iv) Wait about 1 minute and then press the Enter key in terminal emulator and you will see the command prompt "/ #"

      v) Get the ATA-2FXS IP address by typing

      ```
      ifconfig eth0
      ```

      vi) Note this address. The IP address is listed after "inet addr:". You will need it in the next step

   b) Get ATA-2FXS parameters

      i) On the PC, open Internet Explorer

      ii) Type the IP address obtained above in the URL bar; this will open the WebGUI for the ATA-2FXS at its login page (see Figure 5 below)

      iii) Type in admin/admin as user name and password; this will take you to the main page of the WebGUI;

      iv) On the main page, click on "Quick Setup" tab; you will see the screen shown in Figure 6 and Figure 7 below; Note that

         (1) Handset 1 is the FXS line 1.

(2) Handset 2 is the FX2 line 2.

v) Record the following parameters for each of the FXS Lines (shown in figures below)

(1) Outbound Proxy

(2) Proxy

(3) Registrar

(4) User Name

(5) Display Name

You will set these values back after the code upgrade.

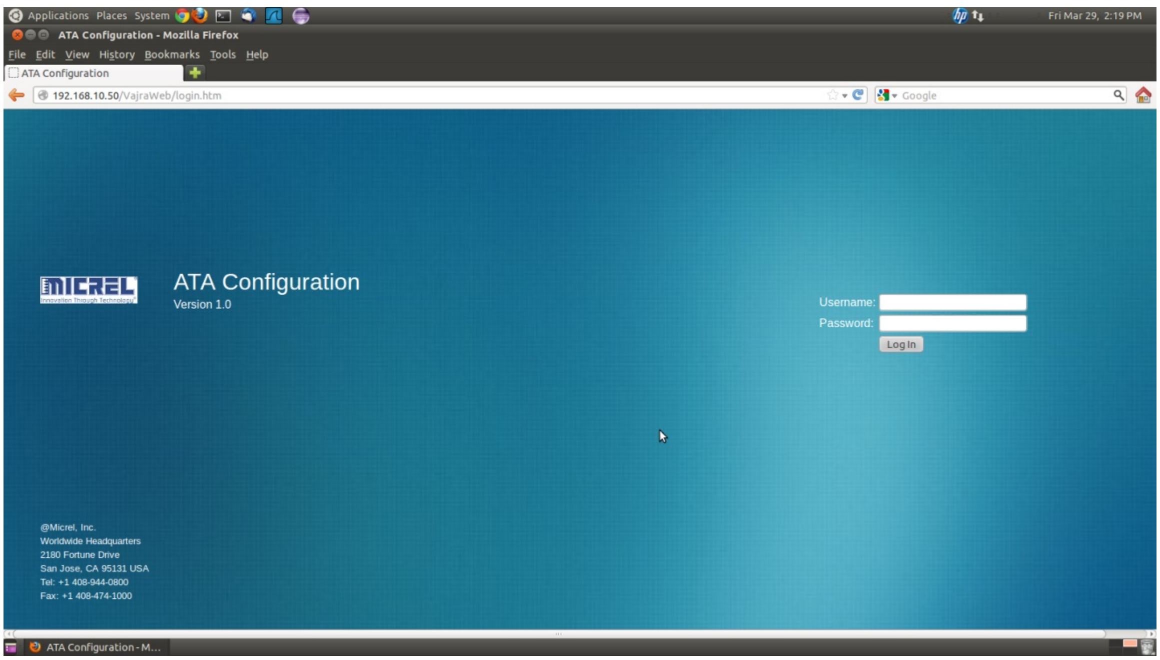


Figure 5: Login Screen

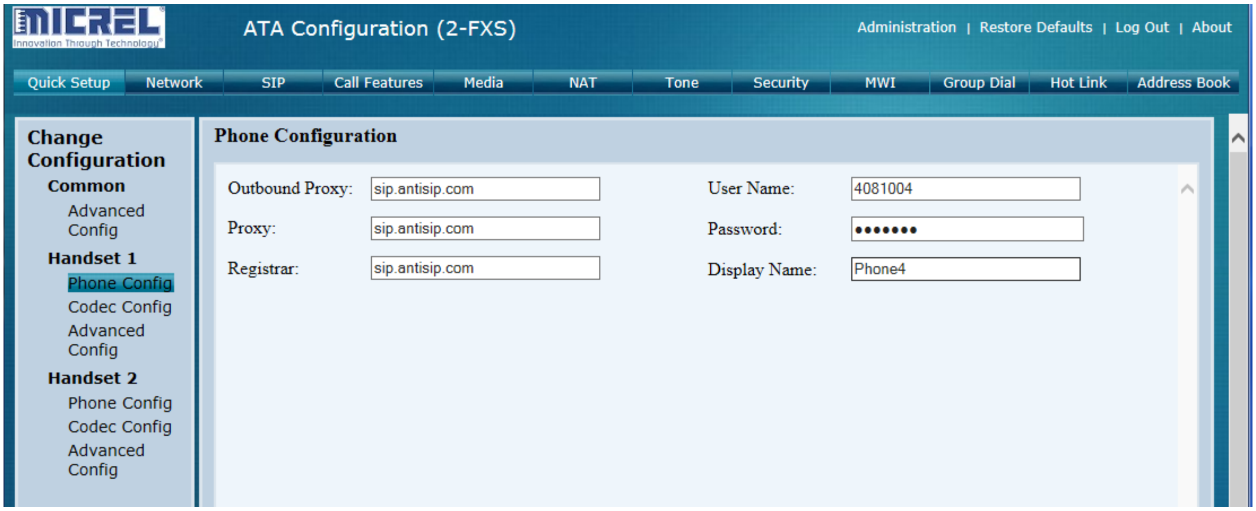


Figure 6: Parameters for ATA-2FXS Line 1

Figure 7: Parameters for ATA-2FXS Line 2

2) Note PC IP address

The PC IP address can be obtained by typing “ipconfig” on the command line prompt. Record this address as it will be used later.

3) Start TFTP Server

The TFTP Server needs to be started, if it is already not.

a) Open the SolarWinds TTP Server

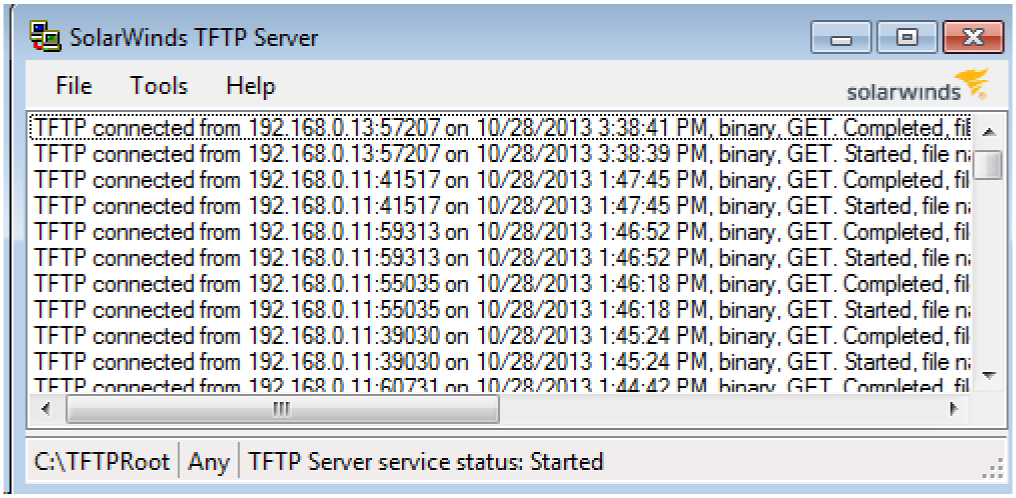


Figure 8: SolarWinds TFTP Server GUI

b) If the TFTP Server Service Status shows “Started”, then the TFTP Server Service is already ON. Skip the next Step

c) If it displays “Stopped” then click on File -> Configure and Click on the “Start” button. Then click on “OK” button. The TFTP Server Service Status should now show “Started”. If it still does not show “Started”, please reinstall the application and/or check your account permission levels on the PC.

4) Configure Environment Variables on ATA-2FXS

a) Power down and power up the board

b) With the Terminal Emulator window in focus, press the “Space” key within 5 seconds of power-up. This will give you access to the boot sector of the ATA-2FXS. You should now see the “#” prompt. If you see additional boot-up messages, you have not pressed the “Space” key in time. Repeat this step.

c) Type the following commands at the “#” prompt.

i) Set IP Address of the ATA-2FXS (obtained from Step 1 above)

```
setenv ipaddr <IPADDRESS OF ATA-2FXS>
```

ii) Set IP Address of the PC (obtained from Step 2 above) on which the TFTP Server is running

```
setenv serverip <IPADDRESS OF PC>
```

iii) Set boot command option

```
setenv bootcmd bootm bf040000
```

iv) Save the confirguration to flash. You should see “Done” when it is finished writing.

```
save
```

v) Reset the board

```
reset
```

d) Press the “Space” key within 5 seconds of power-up. You should now see the “#” prompt. If you see additional boot-up messages, you have not pressed the “Space” key in time. Repeat this step.

11.2.2.3 Obtain New Image

In this step, you will obtain the new Kernel+Application image and put it in the TFTP folder

1) For Micrel-provided image, please download the file from FTP server ….
2) In case, you have built your own image, then get it from the build location
3) Copy the image to the TFTP’s root folder

11.2.2.4 Burn Image

In this step, you will burn the new image into the flash memory.

1) Load the image from the PC to ATA memory with the following command; NOTE: replace INAME with the actual image filename in the TFTP folder:

   ```
   tftp 80200000 INAME
   ```

   When finished, you will see the message “done” and get the prompt back.

2) Since, you are going to write to flash, turn the protection off as follows:

```
sf protect off
```

You will get a message saying flash protection turned off.

3) Erase the previous image from flash using the following command:

```
sf erase 40000 7c0000
```

This step will take ~ 2 minutes. When finished, you will see the message “done” and get the prompt back.

4) Burn the new image to flash from memory to flash. NOTE: use the word “filesize” as shown in the command; it do not mean you need to know the actual file size.

```
sf write 80200000 40000 ${filesize}
```

This step takes ~ 10 minutes. When finished, you will see the message “done” and get the prompt back.

5) Turn flash write protection on, since we are done writing

```
sf protect on
```

At this point, the new image has been burned into the flash.

11.2.2.5 Verify Image

In this step, you will verify that the new image has been burned into flash correctly.

1) Compare image in memory with burned image in flash

```
cmp.b 80200000 bf04000 ${filesize}
```

You will see the message “image is the same”. In case you get an error, redo the Step 2 above.

2) Verify Image Version

   a) Reset the board

   ```
   reset
   ```

   Wait for the board to reboot (~ 1 minute). You will see the “/ #” prompt.

   NOTE: If the bootup fails there could be two potential issues. The image was corrupted in obtaining it and copying it to the TFTP folder. Try to redo this step.

   b) Read version number and date

i) Scroll back on the terminal

ii) Note the following values

(1) Name of the Image --- must be INAME

(2) Date of the Image – must be IDATE

11.2.2.6 Reconfigure ATA Settings

Once the board has rebooted up and the new software is running, you need to set the parameters that we have obtained earlier for the 2 FXS lines.

a) Get the IP Address of ATA-2FXS

i) Connect the PC's serial port to the ATA-2FXS using the null modem serial cable

ii) Open the Terminal emulator (with 115200, 8N1-None settings) – so that you will be able to see the ATA-2FXS boot up messages.

iii) Power up the ATA-2FXS – you will see messages in the terminal window

iv) Wait about 1 minute and then press the Enter key in terminal emulator and you will see the command prompt "/ #"

v) Get the ATA-2FXS IP address by typing

```
ifconfig eth0
```

vi) Note this address. The IP address is listed after "inet addr:". You will need it in the next step

b) Set ATA-2FXS parameters

i) On the PC, open Internet Explorer

ii) Type the IP address obtained above in the URL bar; this will open the WebGUI for the ATA-2FXS at its login page (see Figure 5 above)

iii) Type in admin/admin as user name and password; this will take you to the main page of the WebGUI;

iv) On the main page, click on "Quick Setup" tab; you will see the screen shown in Figure 9 and Figure 10 below; Note that

(1) Handset 1 is the FXS line 1.

(2) Handset 2 is the FX2 line 2.

v) Set the following parameters for each of the FXS Lines (shown in figures below)

(1) Outbound Proxy

(2) Proxy

(3) Registrar

(4) User Name and Password

(5) Display Name

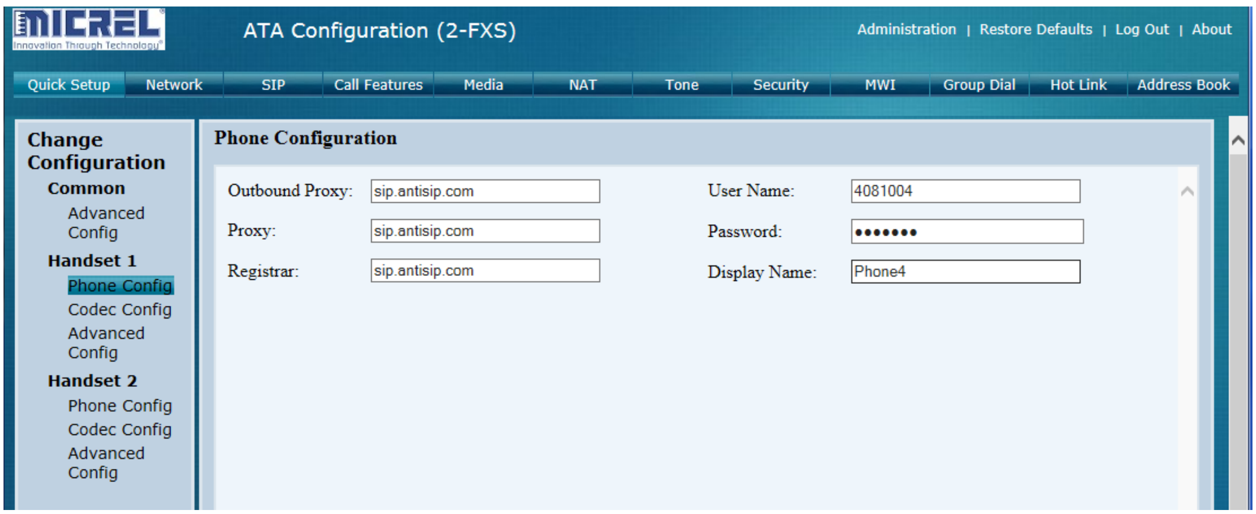


Figure 9: Parameters for ATA-2FXS Line 1

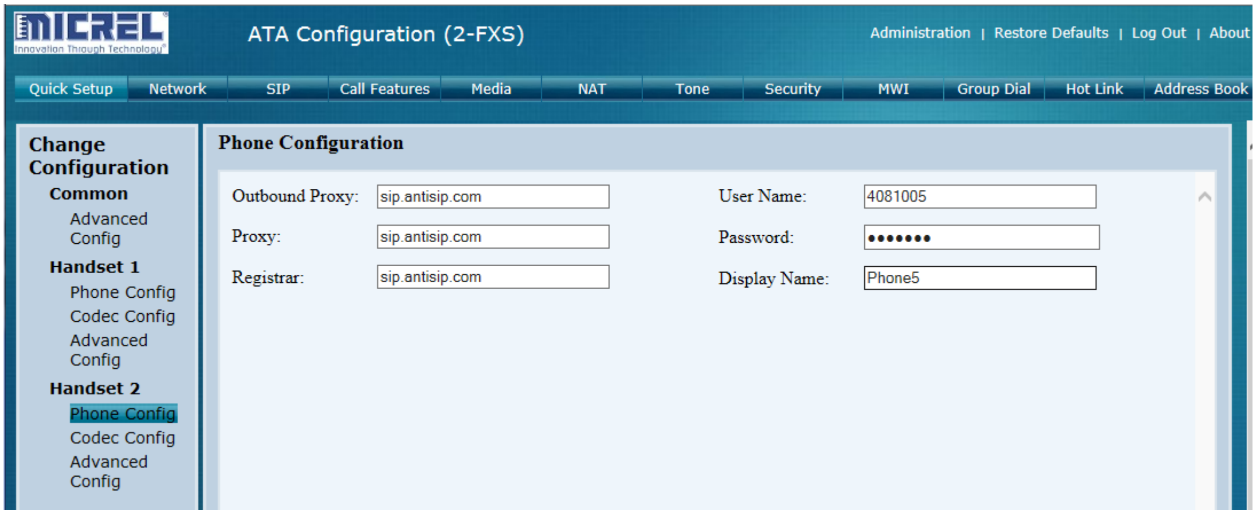


Figure 10: Parameters for ATA-2FXS Line 2

c) Click on Submit button to apply changes to the ATA-2FXS. Wait about 1 minute for the changes to take effect.

You are now done with the Kernel + Application Upgrade!

11.2.3 Update Application

This feature is described in the ATA Evaluation System: Configuration User's Guide; it will be supported in a future release.

11.2.4 Update UBOOT

This section outlines the steps to do an update to the Bootloader software.

NOTE: This section is only to do an update. If the firmware is corrupt and needs to be re-installed or if the board has not yet been programmed, please contact Charles Li (Charles.Li@micrel.com) for information on how to install UBOOT software on the ATA-2FXS.

11.2.4.1 Get Current Configuration Information

1) Note PC IP address

   The PC IP address can be obtained by typing “ipconfig” on the command line prompt. Record this address as it will be used later.

2) Start TFTP Server

   The TFTP Server needs to be started, if it is already not.

   a) Open the SolarWinds TTP Server

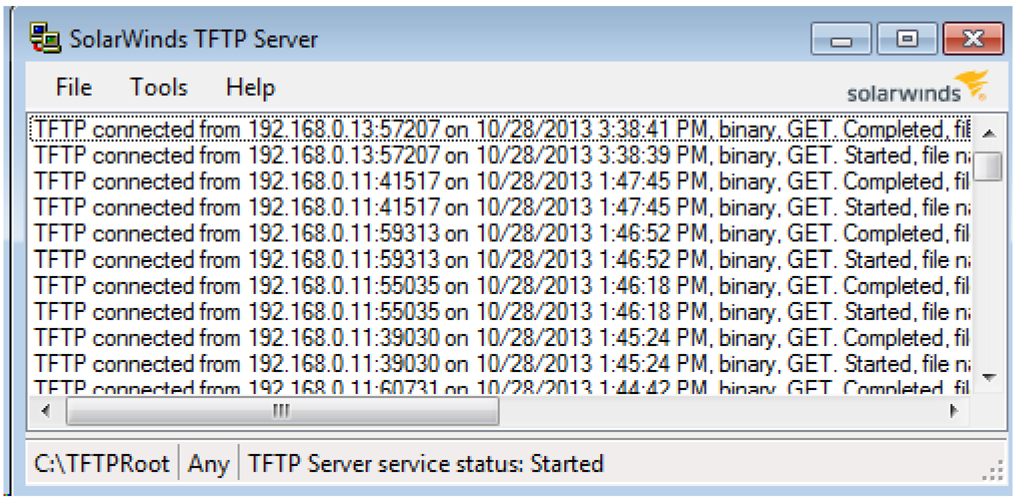


Figure 11: SolarWinds TFTP Server GUI

   b) If the TFTP Server Service Status shows “Started”, then the TFTP Server Service is already ON. Skip the next Step

   c) If it displays “Stopped” then click on File -> Configure and Click on the “Start” button. Then click on “OK” button. The TFTP Server Service Status should now show “Started”. If it still does not show “Started”, please reinstall the application and/or check your account permission levels on the PC.

3) Get the IP Address of ATA-2FXS

a) Connect the PC's serial port to the ATA-2FXS using the null modem serial cable

b) Open the Terminal emulator (with 115200, 8N1-None settings) – so that you will be able to see the ATA-2FXS boot up messages.

c) Power up the ATA-2FXS – you will see messages in the terminal window

d) Wait about 1 minute and then press the Enter key in terminal emulator and you will see the command prompt "/ #"

e) Get the ATA-2FXS IP address by typing

```
ifconfig eth0
```

f) Note this address. The IP address is listed after "inet addr:". You will need it in the next step

4) Get Old UBOOT Version Environment Variables

a) Power down and power up the board

b) With the Terminal Emulator window in focus, press the "Space" key within 5 seconds of power-up. This will give you access to the boot sector of the ATA-2FXS. You should now see the "#" prompt. If you see additional boot-up messages, you have not pressed the "Space" key in time. Repeat this step.

c) Record the UBOOT version and date. This will be used later to verify the new build.

d) Type the following command to see the environment variables

```
print
```

e) This will list all the environment variables. Record the values for the "ethaddr" and "serial#". You will need these in the next step.

5) Erase Old and Set new Environment Variables on ATA-2FXS

a) Type the following commands at the "#" prompt.

i) Erase the memory locations where the Environment variables reside

```
sf erase 30000 10000
```

ii) Reset the ATA-2FXS

```
reset
```

iii) With the Terminal Emulator window in focus, press the “Space” key within 5 seconds of power-up. This will give you access to the boot sector of the ATA-2FXS. You should now see the “#” prompt. If you see additional boot-up messages, you have not pressed the “Space” key in time. Repeat this step.

iv) Set the MAC address of the ATA-2FXS (obtained from Step above)

```
setenv ethaddr <MAC ADDRESS OF ATA-2FXS>
```

v) Set the Serial Number of the ATA-2FXS (obtained from Step above)

```
setenv serial# <MAC ADDRESS OF ATA-2FXS>
```

vi) Set IP Address of the ATA-2FXS (obtained from Step above)

```
setenv ipaddr <IPADDRESS OF ATA-2FXS>
```

vii) Set IP Address of the PC (obtained from Step 2 above) on which the TFTP Server is running

```
setenv serverip <IPADDRESS OF PC>
```

viii) Set boot command option

```
setenv bootcmd bootm bf040000
```

ix) Save the confirguration to flash. You should see “Done” when it is finished writing.

```
save
```

x) Reset the board

```
reset
```

xi) Press the “Space” key within 5 seconds of power-up. You should now see the “#” prompt. If you see additional boot-up messages, you have not pressed the “Space” key in time. Repeat this step.

11.2.4.2 Obtain New Image

In this step, you will obtain the new UBOOT image and put it in the TFTP folder

1) For Micrel-provided image, please download the file from FTP server ….

2) In case, you have built your own image, then get it from the build location

3) Copy the image to the TFTP’s root folder

11.2.4.3 Burn Image

In this step, you will burn the new UBOOT image into the flash memory.

1) Load the image from the PC to ATA memory with the following command; NOTE: replace INAME with the actual image filename in the TFTP folder:

   `tftp 80200000` *`INAME`*

   When finished, you will see the message “done” and get the prompt back.

2) Since, you are going to write to flash, turn the protection off as follows:

   `sf protect off`

   You will get a message saying flash protection turned off.

3) Erase the previous image from flash using the following command:

   `sf erase 0 30000`

   This step will take ~ 1 minute. When finished, you will see the message “done” and get the prompt back.

4) Burn the new image to flash from memory to flash. NOTE: use the word “filesize” as shown in the command; it do not mean you need to know the actual file size.

   `sf write 80200000 0 ${filesize}`

   This step takes ~ 10 minutes. When finished, you will see the message “done” and get the prompt back.

5) Turn flash write protection on, since we are done writing

   `sf protect on`

At this point, **the new image has been burned into the flash!**

11.2.4.4 Verify Image

In this step, you will verify that the new image has been burned into flash correctly.

1) Compare image in memory with burned image in flash

   `cmp.b 80200000 bf04000 ${filesize}`

You will see the message “image is the same”. In case you get an error, redo the Step 2 above.

2) Verify Image Version

   a) Reset the board

      ```
      reset
      ```

   b) Press the “Space” key within 5 seconds of power-up. This will give you access to the boot sector of the ATA-2FXS. You should now see the “#” prompt. If you see additional boot-up messages, you have not pressed the “Space” key in time. Repeat this step.

   c) Read the new UBOOT version number and date. **This should match the new build information**.

12 Revision History

Revision	Date	Summary of Changes
1.0	11/1/13	Initial Release
1.1	11/15/13	Added FXS/FXO