YAMAHA
POWER USER

Motif ES
Sampling in Sequencer Mode

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This article will go over the steps of sampling audio into the sequencer modes of the Motif ES. With expansion capability of 512MB, you have more than 50 minutes of stereo record time at the highest resolution (44.1kHz), that's over 100 minutes of mono recording. This means serious recording can be done. The Motif ES is a **sampler** however and a sampler is different from a hard disk recorder in many ways. They both are tools for music creation via recording audio. However, in a sampler in order to trigger playback you must use MIDI events. Any advantages that a sampler has over a HD recorder are not measured in record time (the HD recorder easily wins that contest every time), but in how you can manipulate and utilize the audio data that you record. It should be mentioned here there is a 'limitation' to the longest single sample that you can take – 6 minutes and 20 seconds. This is a 64MB stereo sample at 44.1kHz. This should not be a limitation really but recognize that you cannot open a mic and record for 50 minutes straight…if that is what you think you need, can we recommend a tape recorder or hard disk recorder like the AW16G!!! This is the reason we stress the point that this is still _sampling_ and the art of sampling is quite a bit different from the art of Hard Disk recording. We will concentrate on recording vocals.

This article assumes that you have properly installed a pair of matching DIMMs and have verified that they are working properly.

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Understanding Audio Sampling: What is it?

"Any track can be MIDI...any track can be audio."
This is a true statement for the Motif ES, however, it is also a trick statement. When you fully understand the way the Motif ES works you will know what I mean. What the statement is trying to say is that there are 16 tracks in the Motif ES sequencer and each track can be used to contain either normal data (playing the keyboard) or can contain audio data (samples). It is also true that Audio samples are assigned to a MIDI note and those MIDI notes when triggered will playback the audio. So the data actually on the track is always MIDI data. The only thing is, this seemingly contradicts our earlier statement. In fact, when you record audio (sample) into the Motif ES, you are creating a special sample Voice. The Integrated Sampling Sequencer, as we will learn, creates a MIDI note-on event that will trigger the playback of that Voice. So all tracks *are* really MIDI tracks – the data that actually goes on a track is always MIDI data, never audio. The audio data, like all sounds in the Motif ES, are stored in *Voices*. All MIDI tracks actually are trigger events that trigger the playback of audio data, be it prepared by Yamaha (preset ROM Voice data) or created by you (sampled audio Voice data).

So while any track can be MIDI and any track can be audio, in fact all tracks are always MIDI and all MIDI tracks always trigger audio. It's a Zen thing! Live it, breathe it in, and absorb it.

The SEQUENCER

It will help to have a basic understanding of how the Motif ES sequencer works. This is so that you can determine which mode will be the best for you to work in. **Song** mode is a linear mode – this means that when the song starts it will count measures in a linear fashion from 001 through to 999. **Pattern** mode is made up of 16 looping Sections – this means that each Section can play over and over. It is non-linear in nature. Pattern mode lets you work on a single section of a song at a time. You might want to concentrate on the Verse or the Chorus section – and have it play over and over while working on it. The key point to understand is that you can move data from Song to Pattern and from Pattern to a Song, as necessary to accomplish what it is you need to accomplish. If you have a *favorite* mode that you use exclusively because you think it is better than the other mode... **grow up**...Learn that these are both tools and that liking a hammer more than a screwdriver is just silly. What if you need a screwdriver? Learn what each tool is for and use it wisely when necessary.

Each session will probably be different and require that you think creatively as to how you want to proceed. What follows here is just a suggestion of a method to proceed. Understand the concepts and then you can develop your own methodology based on what the unit is designed to do. In our first example we will be using the Motif ES to 'overdub' audio to existing MIDI tracks. This is how we speak about it but in fact we will be sampling audio (creating a new VOICE) while monitoring playback of existing data. This is new. Before the Motif sampling was an activity that you did without listening to anything. You, more or less, created an audio sample and then fit your music around that sample. Now we can sample data like you would with a multi-track recorder – that is why we can use the word "overdub". You can sample while you are monitoring (listening back to) recorded tracks. Recording audio is still a linear function and while you can sample in PATTERN mode you will always be creating (naturally) a linear bit of audio. This is not a problem because as we will see a little later in the Motif ES' Pattern sequencer each recording (called a Phrase) can be set to any length you need prior to recording. If you want to record an 8-measure audio track over a 4-measure repeating section, you simply need to open an 8-measure phrase on your audio track. Your backing tracks will repeat 2 times while you sing/play the 8-measure phrase.

'OVERDUBBING' Audio

In this scenario you may have already constructed your basic tracks, drums, bass, keyboard etc. and are adding vocals to the composition. (Of course, you can record anything that you wish; it does not have to be vocals). This will serve as a basic example of how to setup for a session. In this example we have a singer and we intend to do a lead vocal and 3 background/harmony vocal parts. We will build these up one recording at a time – a single singer but we will take 4 passes at this recording. Initially, you may be thinking we need 4 tracks. But as we have been trying to point out there are differences between sampling and Hard Disk recording. In a HD recorder you would need 4 tracks to record this data. But with a sampler, the audio is stored in a Voice. A Voice has 128 places (notes C-2 through G8) on which you can store samples (actually each note can store several samples that can be played simultaneously – but that's another story). So what we will be doing here is setup for our first pass overdubbing to the music tracks recording the lead vocal part on note C3, then we will playback the data on C3 and overdub a backing harmony part on C#3, then playback the lead and backing harmony vocal from C3 and C#3 while we record the second backing part on D3 and so on.

These will all be assigned to a single track…just simply different notes on that single track.

Each note of the sample Voice is autonomous and can be triggered individually (advantage of a sampler). When you think about how samplers store data you really have an unlimited number of storage places, well, more than you can use anyway.
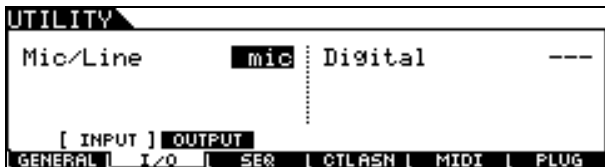
Sampling Audio to a Motif ES Song

Preparation:

Set up the microphone you will be using for recording. A good dynamic microphone is fine for vocals. Do not use an un-powered condenser microphone directly into the Motif ES (it does not have a phantom power supply). You will want to use a good pair of comfortable headphones during the overdubbing session. You should mute your monitor speakers or turn the amp volume completely down so that your speakers do not leak into your sample. This is standard operating procedure in any recording environment. If you are not the singer you can have one set of headphones directly from the Motif ES and the other set plugged into your mixer or you can use a headphone splitter.
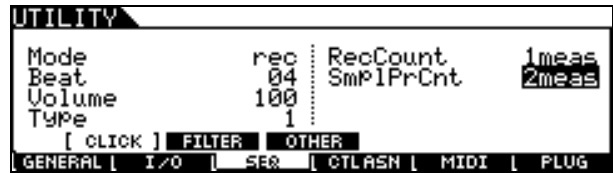
Set the A/D Input
- Press SONG mode and recall your song
- Press UTILITY
- Press F2 I/O
- Press SF1 INPUT – set the MIC/LINE to **MIC**
- Use the Gain control on the back panel of the Motif to raise or lower the input gain.

```
UTILITY
Mic/Line          mic   Digital       ---


         [ INPUT ] OUTPUT
GENERAL |  I/O  |  SEQ  | CTLASN |  MIDI  |  PLUG
```

Set the Metronome (if necessary)
- Press F3 SEQ – because you are in SONG mode when you entered UTILITY, the F3 tab will give you options for the Sequencer set up
- Press SF1 CLICK – set your options for the metronome.

You can select from between 10 different click types. Set the Sample Pre-Count (SmplPrCnt) if you need a count-in prior to sampling. (This is particularly useful in Pattern mode if your vocals begin at the downbeat of the first measure.) It will depend on your music but you may not need the metronome at all (particularly if there is enough music out front to act as a lead in – in which case you can set the MODE to OFF.

```
UTILITY
Mode         rec | RecCount      1meas
Beat          04 | SmplPrCnt     2meas
Volume       100 |
Type           1 |
    [ CLICK ] FILTER   OTHER
GENERAL |  I/O  |  SEQ  | CTLASN |  MIDI  |  PLUG
```

- Recall the SONG and select a blank track for your sample.

Make sure TRACK SELECT is lit and press the track button 1-16 to select a blank track for the vocals.

```
SONG                    Loc1=001 Loc2=001
TMP SCENE| 1 2 3 4|5 6 7 8|9 10 11 12|13 14 15 16
         □ □□□ □□□□ □□□□□ □□□□
01:Dreamflwr              MEAS
Trans:+ 0  E 4/4  ♩135.1  001:01
   ARP1   ARP2   ARP3   ARP4   ARP5
  PLAY  | GROOVE | TRACK            PHRASE | CHAIN
```
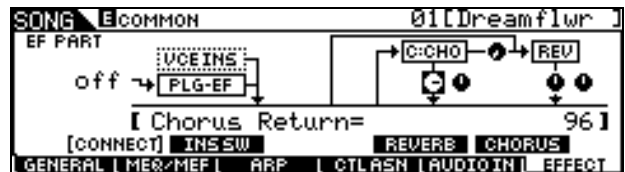
In the above screen Track 16 is selected (track number is clear). Tracks 1, 2, 4, 5, 6, 9, 10, 11, 12, 13, 14, 15 and the Tempo track have data recorded (solid box). A dotted box is an empty track.
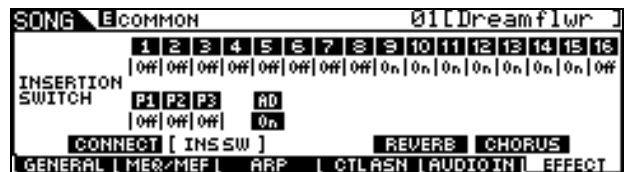
Setup Audio PART
If you want to record with the INSERTION EFFECTS, set that up now.

- Press MIXING
- Press EDIT
- Press COMMON
- Press F6 EFFECT
- Press SF1 CONNECT

```
SONG  ECOMMON            01[Dreamflwr ]
EF PART
            VCEINS         C:CHO  REV
off →  PLG-EF            ↻ ○      ♦ ♦
        [ Chorus Return=          96]
      [CONNECT] INS SW       REVERB  CHORUS
GENERAL | MEQ/MEF |  ARP  | CTLASN |AUDIOIN| EFFECT
```

To turn on the Insertion Effect for the A/D input:
- Press SF2 INS SW (Insert Switch)

```
SONG  ECOMMON            01[Dreamflwr ]
        1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
       |Off|Off|Off|Off|Off|Off|Off|Off|On|On|On|On|On|On|On|Off
INSERTION
SWITCH  P1 P2 P3   AD
       |Off|Off|Off   On
       CONNECT [ INS SW ]    REVERB  CHORUS
GENERAL | MEQ/MEF |  ARP  | CTLASN |AUDIOIN| EFFECT
```

You can turn the AD PART ON if and only if you have an INSERTION SWITCH available. It may be necessary to turn the INSERTION SWITCH OFF temporarily on another Part so that you can record (sample) through it for this operation. Only 8 Insertion Effects can be active at a time. Later

you can turn it back ON for the Part you borrow it from.

Notice that you are placing the INSERTION SWITCH on the AD (analog-to-digital) input PART, which will allow you to record it…You are not routing the track (Track 16 has no data on it yet), besides the track is post the sampler.

Select the INSERTION EFFECT for the AD INPUT:
- Press F5 AUDIO IN
- Press SF2 INS TYPE

```
SONG  COMMON              01[Dreamflwr ]
      1─C:INSA─┐
      2─C:INSB─┘
      [InsA Type= ---:Thru        ]
      OUTPUT [INSTYPE]
GENERAL MEQ/MEF  ARP  CTLASN AUDIOIN EFFECT
```
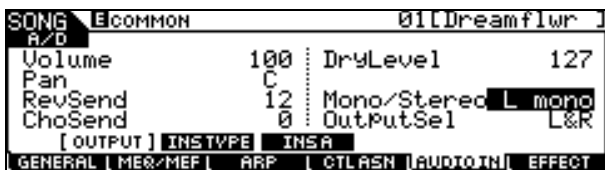
Here you have the dual processors with 116 algorithms each. These effects applied here will be recorded (sampled) to the track. You have to make a decision whether or not you want to do this now or later. It is "what-you-hear-is-what-you-get". You can select from among Compressors, Reverbs, Chorus, Delays, etc., etc.

```
SONG  COMMON              01[Dreamflwr ]
      1─C:INSA─┐
      2─C:INSB─┘
      [InsA Type= DYN:Compressor   ]
      OUTPUT [INSTYPE] INSA
GENERAL MEQ/MEF  ARP  CTLASN AUDIOIN EFFECT
```

Above INS_A is set to the DYNAMICS category and a Compressor is selected.

- Press SF3 INS_A (and SF4 INS_B, if selected) to edit the selected Insertion Effect parameters.[1]
- Press SF1 OUTPUT - Set the parameters for the AD PART, including the pan, output level, Mono/Stereo, etc.

```
SONG  COMMON              01[Dreamflwr ]
A/D
Volume        100  DryLevel       127
Pan            C
RevSend        12  Mono/Stereo  L mono
ChoSend         0  OutputSel       L&R
      [OUTPUT] INSTYPE  INSA
GENERAL MEQ/MEF  ARP  CTLASN AUDIOIN EFFECT
```

- Set the Mono/Stereo to **"L_mono"** to correspond to the jack you are using on the back panel of the Motif ES for the microphone input.

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[1] See page 9 on "Setting up a Compressor" for details on working with dynamics processing.

- Increase the Volume as necessary for comfort.
- The Rev_Send and Cho_Send cannot be recorded. These will not be recorded (sampled). If you want to record with Reverb you can use one of the two INSERTION EFFECTS to select an appropriate reverb. Technically, we are patching the INSERTION EFFECT on the input stage of the AD prior to the recorder (sampler). The Reverb and Chorus processors are *post* the recorder and therefore not able to be sampled (or heard) when we enter the sampling mode.
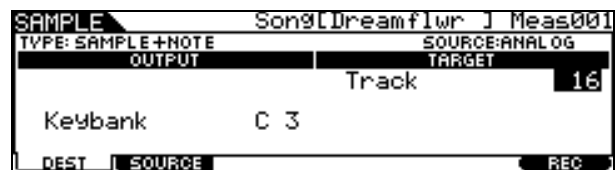
Setup the Sampler
- Press INTEGRATED SAMPLING
- Press F2 Source – here we will tell the unit where the sample is coming from and what type of sampling we are going to be doing.
- Set the Sample TYPE to **'sample+note'**
- SOURCE is **A/D**
- Set the Mono/Stereo to **'L+Rmono'**

```
SAMPLE       Song[Dreamflwr ] Meas001
Type    sample+note  Mono/Stereo L+Rmono
Source         A/D   Frequency      44.1k
Next           off

  DEST  SOURCE                         REC
```

Note that MONO/STEREO is set to **'L+Rmono'** – this will allow you to monitor the input in the Center when your source is mono (microphone).

- Press F1 Destination

```
SAMPLE       Song[Dreamflwr ] Meas001
TYPE: SAMPLE+NOTE          SOURCE:ANALOG
    OUTPUT                 TARGET
                   Track              16

Keybank        C 3

  DEST  SOURCE                         REC
```

Here you can review that you have assigned the Target Track to 16 and since we have selected "Sample+Note" as the sampling TYPE you can target a *specific* note for the MIDI trigger event, **C3**. Sample+Note means literally that the sample will be recorded and that a note-on event will be created to trigger playback of that audio. The duration (gate time) and placement of the note will be determined by the TRIGGER Mode selected in our next setup step.

- Press F6 REC – To setup for the actual sample

You can test the microphone and see the bar graph that represents the record level. The object is to get as much level as possible without clipping. You will see an icon of lightning bolts

when you are clipping. The more level the better – short of distortion. Adjust the GAIN control on the back panel for more or less level. The RecMonitor parameter is set for you listening comfort.

• Select a TRIGGER MODE:
Level – sampling begins when audio exceeds a particular setting value. Sampling ends manually by pressing F6 STOP
Measure – sample begins when the sequencer reaches the Punch-in point and ends when it reaches the Punch-out point
Manual – sampling begins when you press F6 REC and ends when you press F6 STOP
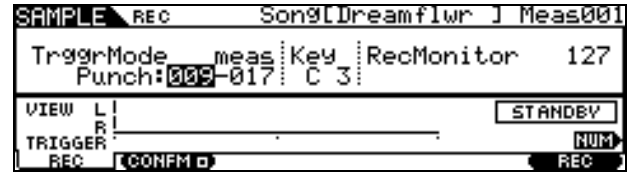
How to choose which TRIGGER MODE...
The Trigger mode that you select will have to be determined by what you want to accomplish. The advantages of one over the other will be determined by your particular project.

Level would be useful if you are not sure when the audio is to begin. As soon as the signal passes the pre-determined level, sampling begins. There is a 100ms pre-record buffer so that you actually will have 100ms prior to the level being exceeded. This allows you record additional information and you can edit the front end in case the attack is missing. In general, the LEVEL Trigger mode is used when you are recording data from a typical audio sampling CD where you cue the source to a particular track. Sampling begins automatically. Press F6 REC to arm the sampler, and then press play on the CD player, the Motif ES screen will read, "Waiting" until the level reaches the trigger point – sampling begins. You manually stop it by pressing F6 STOP. The note-on is created from the clock pulse that sampling begins and the duration will continue through to the point where you press STOP.
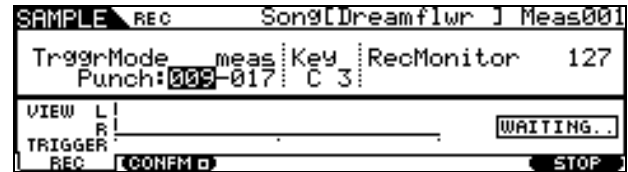
Meas or Measure is used when you know the punch-in and out points by measure. This is useful for most overdub type samples such as vocals or guitars, etc. In this method the sequencer controls the punch-in and the punch-out points. This method is fully automated. Press F6 REC, the screen will read "Waiting" until the sequencer is started and reaches the punch-in measure. The note-on is created from the punch-in point and the duration will be the length between the in and out points.

Manual is used when you want to control the start and stop of the sample yourself. Sampling begins when you press F6 REC and ends when you press F6 STOP. The note on is created from the moment that you hit REC and the duration will be the length between this point and when you hit STOP.

Since we are sampling to the sequencer, we will use the MEASURE trigger type. Determine at what measure you want to punch-in and punch-out.

```
SAMPLE  REC       Song[Dreamflwr ] Meas001
TrggrMode   meas Key RecMonitor    127
   Punch:009-017  C 3
VIEW  L|                              STANDBY
      R|
TRIGGER|                                 NUM
   REC  [CONFM ]                        ( REC )
```

In the screen shot above (Standby), the Trigger Mode is **Measure**, we are set to punch in at the top of measure **009** and punch out at the top of measure **017** ...Press F6... "Waiting..." appears. The current measure reads out in the upper right corner. Use the [>>] FF button to locate a comfortable start point prior to the punch-in point.

```
SAMPLE  REC       Song[Dreamflwr ] Meas001
TrggrMode   meas Key RecMonitor    127
   Punch:009-017  C 3
VIEW  L|
      R|                              WAITING..
TRIGGER|
   REC  [CONFM ]                        ( STOP )
```

...then start the sequencer playback, and the sampling will begin, automatically, when the sequencer passes measure 009 and will punch out at the beginning of measure 017. You can start playback from some measure ahead of the punch-in point. This will allow the singer to get oriented to the feel of the music prior to singing.

Playback the Sample Voice: Confirm
When sampling is complete you will be given the opportunity to AUDITION the recording via the F3 button.
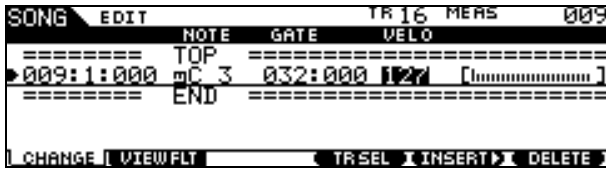• Press and hold it until to hear your recording playback (isolated).
• Press F5 OK to accept the results or F4 CANCEL to discard and try again.
When you press OK you have created a special User Sample Voice that is 'local' to the current Song. It will reside in bank 63/50 and as this was the first sample in this Song it will be designated as User Sample Voice 1. The Integrated Sampling Sequencer will have created a note-on on track 16.

The default Velocity is 100. The default Voice Volume is 100. If your sample is not loud enough (quite possible as a raw sample) you can increase the Velocity and the Volume:
• Press SONG
• Press EDIT to view the Event List
• Use the cursor arrows to navigate this screen
• Set the Velocity to 127 (shown highlighted below)

- Press ENTER

```
SONG   EDIT              TR 16  MEAS     009
                NOTE     GATE      VELO
========  TOP   ==============================
►009:1:000 mC 3  032:000 127 [␣␣␣␣␣␣␣␣␣␣␣␣␣]
========  END   ==============================

 CHANGE  VIEWFLT          TRSEL  INSERT  DELETE
```

- Press MIXING
- Press F1 VOL/PAN. Highlight the fader.
- Set the Volume to 127 or as necessary
- Press STORE/ ENTER/ YES to store the MIX

Note in the screen above a note-on event was created on Track 16 at measure 009, beat 1, clock tick 000; the note is C3 (middle C); the duration or gate is 32 beats and 000 clocks (that's 8 measures of 4/4 time) and the Velocity has been set to 127. "TOP" denotes the beginning of this track and "END" denotes the end of that track – so this note-on event is currently the only event on that track.

SETUP for Overdubbing additional data
We can now setup to record our next vocal sample to the next highest key (C#3). Using the same setup the ISS will now playback the track, including the vocal on C3, as we overdub a backing vocal on C#3. Do not worry, it will not erase or disturb in the least the audio you sampled on C3…it will create a new sample that can be triggered separately, and even played back a different volume. It will be an independent sample, with an independent note-on event.

- Press INTEGRATED SAMPLING
- Press F1 DEST – set the Target as **C#3**; Track 16
- Press F2 SOURCE – to verify the setup (it will be the same)
- Press F6 REC – to arm the sampler. We will use exactly the same setup
- Press F6 REC – to enter Standby. Change the Trigger Mode, Measures, etc. as necessary to record this next track.
- Press F6 REC – to arm the sampler
- Press [>] to start playback of the sequencer

Overdubs can be executed, as outlined above, for as many passes as you want. There are 128 MIDI notes per Channel. The samples go to the same track but a different key. When doing the setup you will see the "NEXT" parameter – this can automatically advance you to the next higher chromatic note and return you to the sample 'Standby' screen.
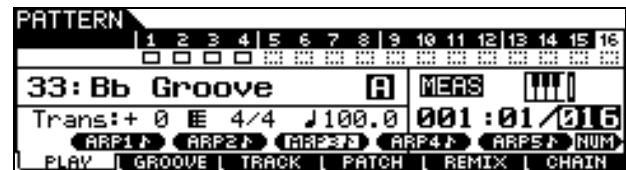
The way you would group your recording is according to how you want to work with them. Vocals of the same singer would naturally share the same EQ and Effects settings and are likely to be muted or unmated together. This makes them candidates for recording to the same track. You wouldn't want to put things on the same track that you want to EQ or effect differently. You will have to decide what is appropriate for your particular session on a per session basis.

Sampling Audio to a Motif ES Pattern
Nothing prevents you from sampling into a Pattern Section. However, the difference between Song mode (linear) and Pattern (cycling Sections) will cause a few differences in how you approach the task. Fundamentally, the setup is the same. The differences will be in how sampling starts and ends. When setting up for Pattern mode sampling you will need to change the Phrase Length prior to sampling. In fact you may determine that sampling vocals might be better done *after* you convert the Pattern Sections to a CHAIN and the Chain to a SONG. This could be due to things like pickups (where the audio begins prior to the downbeat) or overhangs (where a note is held into the next section). However, if you do determine that you want to sample into a Pattern please understand that sampling is linear recording – i.e., sampling will begin and end while the Pattern plays. The length will be determined by the Length parameter.

To program the Length of any Phrase in Pattern mode:
- Press PATTERN to view the main Pattern screen
- Move the cursor to the right and highlight the Length parameter (shown highlighted below)

```
PATTERN
         |1 2 3 4|5 6 7 8|9 10 11 12|13 14 15 16
         □□□□□ ▫▫▫▫ ▫▫▫▫ ▫▫▫▫ ▫▫▫▫
 33: Bb Groove        A  MEAS  ▏▎▍▌
 Trans:+ 0  E 4/4  ♩100.0  001:01/016
      ARP1♪  ARP2♪  ARP3♪  ARP4♪  ARP5♪ NUM
 PLAY   GROOVE  TRACK  PATCH  REMIX  CHAIN
```

In the screen shot above the highlight is on the LENGTH parameter: 016. The numbers translate into 001:01/016 = Current position is Measure 001, beat 01 out of 016 measures total. By setting the length prior to recording you are setting the end point of the phrase. This is where the sequencer will automatically punch out. Remember this setting is per track phrase. Each track's phrase can be a different length as necessary. In other words, you could have a 1-measure drum groove, a 2-measure synth bass part, a 4-measure keyboard part, and then open

up a 16-measure vocal phrase (any length up to 256 measures).

Set the A/D Input
- Press PATTERN mode and recall your pattern
- Press UTILITY
- Press F2 A/D
- Press SF1 INPUT – set the MIC/LINE to **MIC**

Set the Metronome (if necessary)
- Press F3 SEQ – because you are in PATTERN mode when you entered UTILITY, the F3 tab will give you options for the Sequencer set up
- Press SF1 CLICK – set your options for the metronome.

Set the Sample Pre-Count (SmplPrCnt) if you need a count-in prior to sampling. (This is particularly useful in Pattern mode if your vocals begin at the downbeat of the first measure.) It will depend on your music but you may not need the metronome at all – in which case you can set the MODE to OFF.

- Recall the PATTERN and select a blank track for your sample.

Make sure TRACK SELECT is lit and press the track button 1-16 to select a blank track for the vocals.

In the above screen Track 16 is selected (track number is clear). Tracks 1, 2, 3 and 4 have data recorded (solid box). A dotted box is an empty track.
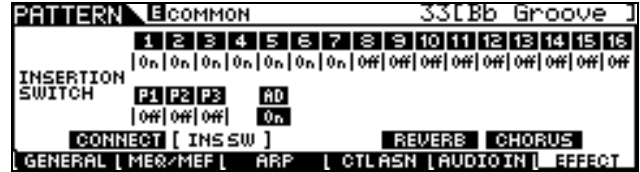
Setup Audio PART
If you want to record with the INSERTION EFFECTS, set that up now.

- Press MIXING
- Press EDIT
- Press COMMON
- Press F6 EFFECT
- Press SF1 CONNECT

To turn on the Insertion Effect for the A/D input:
- Press SF2 INS SW (Insert Switch)

You can turn the AD PART ON if and only if you have an INSERTION SWITCH available. It may be necessary to turn the INSERTION SWITCH OFF temporarily on another Part so that you can record (sample) through it for this operation. Only 8 Insertion Effects can be active at a time. Later you can turn it back ON for the Part you borrow it from.

Notice that you are placing the INSERTION SWITCH on the AD (analog-to-digital) input PART, which will allow you to record it…not on the track (Track 16 which has no data on it yet).

Select the INSERTION EFFECT for the AD INPUT:
- Press F5 AUDIO IN
- Press SF2 INS TYPE

Here you have the dual processors with 116 algorithms each. These effects applied here will be recorded (sampled) to the track. You have to make a decision whether or not you want to do this now or later. It is "what-you-hear-is-what-you-get". You can select from among Compressors, Reverbs, Chorus, Delays, etc., etc.
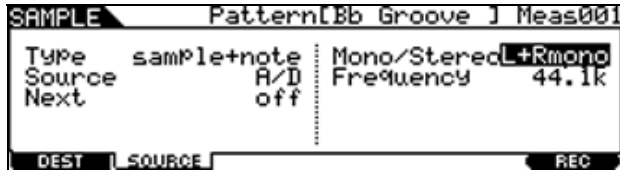
If you want to record with both Reverb and Compressor you might want to set INSERT B to a Reverb Hall.

- Press SF1 OUTPUT - Set the parameters for the AD PART, including the sends, output level, Mono/Stereo, etc.
- Set the Mono/Stereo to **"L_mono"** to correspond to the jack you are using on the back panel of the Motif ES for the microphone input.
- Increase the Volume to **127**.
- The Rev_Send and Cho_Send cannot be recorded. These will not be recorded (sampled). If you want to record with Reverb you can use one of the two INSERTION EFFECTS to select an appropriate reverb. Technically, we are patching the INSERTION EFFECT on the input stage of the AD prior to

the recorder (sampler). The Reverb and Chorus processors are post the recorder and therefore not able to be sampled when applied to a live input.

Setup the Sampler
- Press INTEGRATED SAMPLING
- Press F2 Source
- Set the Sample TYPE to **'sample+note'**
- SOURCE is **A/D**
- Set the Mono/Stereo to **'L+Rmono'**

```
SAMPLE\        Pattern[Bb Groove ] Meas001
┌─────────────────────────────────────────┐
│ Type    sample+note│Mono/Stereo L+Rmono  │
│ Source          A/D│Frequency      44.1k │
│ Next            off │                     │
│                    :                      │
│                    :                      │
└─────────────────────────────────────────┘
 DEST  SOURCE                          REC
```
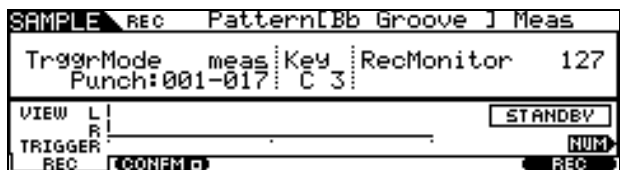
Set the MONO/STEREO to **'L+Rmono'** – this will allow you to monitor the input in the Center when your source is mono (microphone).

- Press F1 Destination

Here you can review that you have assigned the Target Track to 16 and since we have selected "Sample+Note" as the sampling TYPE you can target a *specific* note for the MIDI trigger event, **C3**. Sample+Note means literally that the sample will be recorded and that a note-on event will be created to trigger playback of that audio. The duration (gate time) and placement of the note will be determined by the TRIGGER Mode selected in our next setup step.

- Press F6 REC – To setup for the actual sample

You can test the microphone and see the bar graph that represents the record level. The object is to get as much level as possible without clipping. You will see an icon of lightning bolts when you are clipping. The more level the better – short of distortion. Adjust the GAIN control on the back panel for more or less level.
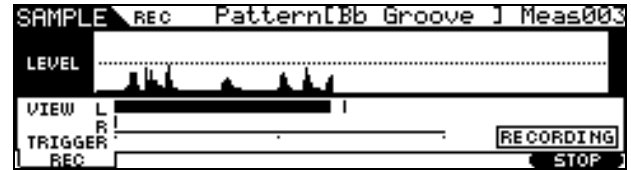
Select Measure as the TRIGGER MODE and set the Punch-in and Punch-out points.

```
SAMPLE\ REC   Pattern[Bb Groove ] Meas
┌─────────────────────────────────────────┐
│ TrggrMode    meas│Key│RecMonitor    127 │
│   Punch:001-017  │C 3│                   │
│ VIEW  L│                                 │
│       R│                      STANDBY    │
│ TRIGGER│          .         .      NUM   │
│  REC   CONFM                        REC  │
└─────────────────────────────────────────┘
```
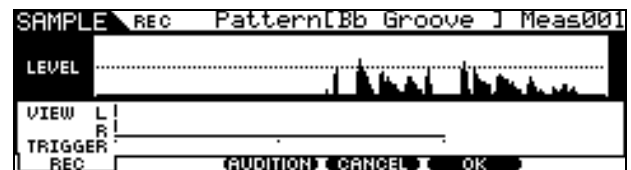
In the example above the Trigger Mode is set to Measure. Notice that the punch-out point is measure 17. This denotes that it will complete 16 measures of recording. In this example recording

from measure 1 will mean that the Sample Pre-Count will be necessary. In the UTILITY/ F3 SEQ/ SF1 CLICK you would set the SmplPrCnt to count in for you.
- Press F6 REC to see "Waiting.."
- Press [>] To begin sequencer playback

```
SAMPLE\ REC   Pattern[Bb Groove ] Meas003
┌─────────────────────────────────────────┐
│ LEVEL  ·····░░░░···················       │
│                                           │
│ VIEW  L ████████████████ |               │
│       R                                   │
│ TRIGGER          .         .   RECORDING  │
│  REC                               STOP   │
└─────────────────────────────────────────┘
```

When the sequencer reaches the end of measure 16 sampling will stop automatically.

```
SAMPLE\ REC   Pattern[Bb Groove ] Meas001
┌─────────────────────────────────────────┐
│ LEVEL  ·····························       │
│                                           │
│ VIEW  L                                   │
│       R     ░░▓▓░░░░  ░░▓▓▓░░░░           │
│ TRIGGER          .         .              │
│  REC        AUDITION  CANCEL    OK        │
└─────────────────────────────────────────┘
```
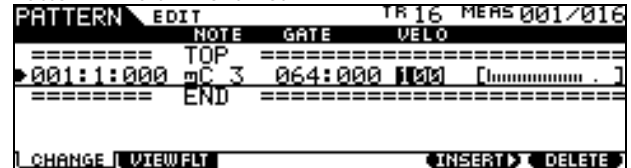
You will be given the chance to AUDITION the recording by pressing F3, CANCEL F4 and/or keep the recording F5 OK.

If you press F5 OK the Integrated Sampling Sequencer will create a new sample Voice (Bank 63/50, Voice number 1) and a MIDI note-on will be placed in track 16 at the punch-in point to trigger playback, with a duration (gate time) to complete the playback of the audio:

Mixing > F2 Voice: Mix view
```
PATTERN\ MIX    Voice=001[--:Wave0001   ]
 PART   1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
        DR BA KB GT AP AP AP AP AP AP AP AP AP AP AP --
VOICENUM  1 105 69 12                          1
BANK MSB 63 63 63 63 63 63 63 63 63 63 63 63 63 63 63 63
BANK LSB 32  0  0  0  0  0  0  0  0  0  0  0  0  0  0 50
         PD P.2 Us1 P.1 P.1 P.1 P.1 P.1 P.1 P.1 P.1 P.1 P.1 P.1 P.1 SP
 VOL/PAN  VOICE   EFFECT   EFSEND            PLG1-3
```

Pattern > Edit: Event List
```
PATTERN\ EDIT             TR16  MEAS 001/016
             NOTE    GATE    VELO
======== TOP ===========================
►001:1:000 mC 3   064:000 100  [|||||||||| . ]
======== END ===========================

 CHANGE  VIEWFLT              INSERT  DELETE
```

Main Pattern screen: Note: Wave icon on track 16
```
PATTERN\
        1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
        □ □ □ □                              ▨
33:Bb Groove        A  MEAS  ▮▮▮▮
Trans:+ 0  E 4/4  ♩100.0  001:01/016
   ARP1  ARP2  ARP3  ARP4  ARP5  NUM
 PLAY  GROOVE  TRACK  PATCH  REMIX  CHAIN
```
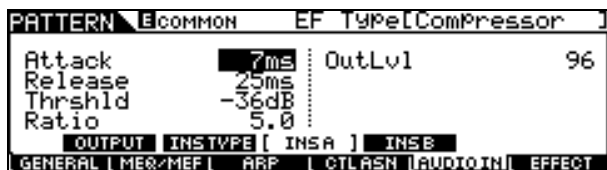
Overdubbing can continue as illustrated in the SONG mode section.

Setting up a Compressor

The use of a compressor can separate your recording from a mediocre sample and move it into the realm of greatness. Normalizing samples is what many users wind up doing because they do not have the professional tools of the trade. "Normalizing" is a term used in the sampling world for bringing the loudest peak of the sample up to theoretical maximum – it also brings up the noise floor as well. Yes it makes your recording perceptibly louder but it does not give your recording "presence". This is why recording your sample with a professional tool like a compressor can give you much better sounding results.

When recording vocals in any professional recoding studio, a processor called a compressor is often used. A compressor is in a category of tools called Dynamics Processing. And what a Dynamic Processor does is help you get the best recording level possible without overloading the medium. Compressors are technically "Leveling Amplifiers". A compressor is defined as an amplifier where the more you put in, the less you get out. It has a point called the *threshold* and when signal exceeds the threshold the compressor will automatically reduce the level of the input (Gain Reduction). The result of all this is that it automatically prevents overload by bringing the loudest peaks down so that they are closer to the softest portions of the signal. This is referred to as reducing the *dynamic range* (basically the range from the softest signal that the audio source makes to the loudest signal that the audio source will make). Use it when you need a track to stand out – what musicians/engineers refer to as 'have presence'…this is when artful use of a compressor is called for. Knowing how to set up a compressor is an art. One setting will not work all the time (so memorizing one setting will not do you much good). The same singer from song to song might require a different setting. Knowing the goal and the parameters and what you are after will help you use the compressor wisely. The parameters that you have control over are as follows:

```
PATTERN ▟ⒺCOMMON       EF Type[Compressor    ]
Attack        ▊7ms  OutLvl              96
Release        25ms
Thrshld       -36dB
Ratio          5.0
      OUTPUT INSTYPE [ INSA ] INSB
 GENERAL MEQ/MEF   ARP   CTLASN AUDIOIN EFFECT
```

Attack - Attack controls the first few milliseconds of the signal (1ms-40ms). It will allow the signal to pass before the compressor engages. Percussive instruments are a rather fast attack with a big transient peak at the attack, as opposed to a vocal that might sometimes be fast or slow - you will want to make attack setting faster (lower the number) for items that are sharp attacks. A transient peak is the stick hitting the skin in a drum sound. Always think of all sounds as a two-part entity: "cause and affect". The stick hitting (transient peak) and then the response of the drum (the body of the sound) are the two parts of a drum sound. In vocals you have the noise component that starts a word or lyric and the vowel component that is the sustained portion.

Release - Release time will be dependent on your tempo as well as the type of sound (10ms-680ms). This is the time it takes for the compressor to 'let go' of the signal. Because a snare does not really 'sustain' in the sense of like a guitar or vocal this is very subjective and passes probably without much notice - guitar players use long release times to get a *perceived* increased sustain. The picked attack is the transient peak and has lots of energy, the sound then dies a bit (called the initial decay) to the sustain level (main body of the sound). If you compress the big (transient) peak of energy on the attack the whole signal seems to remain at the same volume longer…and is perceived as longer sustain. The more you compress and the longer the release the longer the perceived sustain. With vocals you will use your ears and common sense to direct you about release time.

Threshold – Threshold is a level setting and when signal goes above this point the compressor will start to reduce the dynamic range - bringing down the transient peak energy closer to the body of the sound (-48dB~-6dB). Threshold is a critical setting because signal below the threshold level is unaffected by the compressor. You want to set the threshold to reduce just the loudest peaks, not the entire signal…only the loudest peaks. Raise the Threshold to an appropriate level for the input signal.

Ratio – Ratio is the degree that the compressor is working. The ratios are always to 1. Ratios of 1.5, 2.0, 3.0, and 5.0 are appropriate for vocals, with 5:1 a very severe compression. 1.0 can be used as a straight amplifier no compression occurs. If for example the signal were to try to go as high as -2dB and you had a Threshold set to engage whenever the signal reached the –4dB point and you had a Ratio set to 2:1, then the signal would only reach –3dB. It would be reduced by a 2:1 ratio; that is for every 2dB it want to increase past the threshold the compressor allows 1dB out. If that same signal got even louder at the source because the singer is now belting it out, and would go all the way to +2dB (and distortion), the

2:1 ratio would kick in when the audio signal tripped past the threshold at −4dB it would only be allowed to increase half that distance to −1dB. Instead of traveling the 6dB to +2dB it can only move half the distance - as high as −1dB. The difference between a Compressor and a Limiter is one of degree (10:1 and above are limiting). You are stepping on it pretty good - limiting the maximum. In order for the signal to reach the same −1dB level with a 10:1 ratio, the signal would have to be strong enough to go to +26dB. That's 30dB above the threshold (yikes).

Output Level is so that you can return the now compressed dynamic range to its proper balance within the mix (0-127). The idea with a compressor/limiter is you are making the soft level closer to the loud level (i.e., compressing the dynamic range) so the output is **very** important to restore the balance (the level within the whole mix). Think of a singer who whispers over a big band track - whispers are soft. Then the singer belts out at the top of their lungs – that' the loud. A compressor would bring the loudest signal down closer to the whisper's level… compressing the dynamic range…then the all important OUTPUT would let you raise the overall signal back up. So the whisper will sit and be heard in context of the mix yet the shout will not clip the levels! that's the purpose of the compressor/limiter.

The **over** use of a compressor will make a singer sound like they have asthma because the sound of breathing in and out (normally soft) will be too close to the actual vocal sound. The compressor will ignore the soft sounds (as they are below the threshold) and it will compress the louder sounds. Be careful, nothing is worse than over-compression. Engineers talk about the compressor pumping or breathing when you are dangerously close to this point…less is more in this scenario. The rule is: "By the time you hear the compression, you have too much". No discussion of compression would be complete without a warning…do not compress everything and use it with good sense. What is more important than LOUD in any good mix is the balance between soft and loud. If there is only loud in your mix, then your mix will become difficult to listen to and fatiguing to the ear. If your mix has dynamics (both soft and loud) then the ear will treat it like candy and love moments that have meaning and impact. Test both loud and soft dynamics when setting the compressor. Remember you only want to reduce the loudest peaks. The −36 default setting is probably way too much compression – raise the threshold to a more reasonable amount.

Making your sample fit the Mix
The next thing to do is to make sure your recording can be heard in the mix balanced with the other Parts. You can, as mentioned earlier, raise the Volume and increase the Velocity of the trigger note – this will *significantly* affect the overall impact. You can also apply PART EQ, effects, pan, etc. to your recording. It now resides in the special local bank 63/50. The sample will be saved with the Song or with the Pattern it was created in. This sample goes to a special folder in the Motif ES file architecture and can be loaded individually (File Type: SAMPLE VOICE) from within any ALL SONG (.w7s), ALL PATTERN (.w7p) or ALL data (.w7a) file.

Creating a USER VOICE
As an alternative method you can create a USER Voice from your recording. This will place it in bank 63/8 or 63/9 (User 1 or User 2) and will allow you to fully edit the sound like any other Voice in the Motif ES.
* Go to Voice mode and initialize a new USER VOICE
* Press JOB/ F1 INIT/ ENTER/ YES
* Press EDIT.
* Press Track 1 to select Element 1
* Press F1 OSC/ SF1 WAVE point the Voice to your USER wave

Now you can bring the full editing power of the Voice mode to your recording. Many users wonder why their sample is not as loud as the other Parts of the mix…well the answer is simple. The other Parts of the mix contain *programmed* Voices. All you did was make a raw sample – of course it is not as loud, you have not applied any program functions to it. By making your recording a user VOICE you can now fully edit it. Located at **F6** in the view above is the Element **EQ** which can be configured as a single band parametric, a 2-band EQ or as a straight +6, +12 or +18dB boost.

* Store it to a USER location – it will be in bank 63/8 (USER 1) or 63/9 (USER 2).

Creating a MIX VOICE
Once you have stored the sample recording as a USER bank Voice be sure you point your SONG or PATTERN mix to that User Voice. STORE your mix.

```
PATTERN  MIX    Voice=001[--:VocalTrack]
PART     1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
         DR BA KB GT AP AP AP AP AP AP AP AP AP AP AP --
VOICE NUM  1 105 57 12  1  1  1  1  1  1  1  1  1  1  1  1
BANK MSB  63 63 63 63 63 63 63 63 63 63 63 63 63 63 63 63
BANK LSB  32  1  0  8  0  0  0  0  0  0  0  0  0  0  0  8
         PD P-2 P-1 Us1 P-1 P-1 P-1 P-1 P-1 P-1 P-1 P-1 P-1 P-1 P-1 Us1
VOL/PAN  VOICE [ EFFECT ] EF SEND [ VCE ED ] PLG1-3
```

- Press MIXING
- Press EDIT
- Press the Track that contains your new USER VOICE
- Press F5 VOICE ED

This will drop you into Voice edit, now you can store the Voice to a special "local" location within the MIX called the MIX VOICE (bank 63/60). Each of the 16 PARTS of a mix can store 1 MIX VOICE.

- Store the Voice to the MIX VOICE location

```
PATTERN  VOICE STORE Part16[--:VocalTrack]

Mixing Voice Store
         MIXV:016[ :          ]


                    PRESS [ENTER] TO STORE.
```

The advantage of placing your User Voice in the 63/60 MIX VOICE bank is that – being "local" means it will always load when you load the SONG or PATTERN that it is in. The MIX VOICE bank can hold a single Normal (non-Drum Kit, non-PLG) Voice per each Part 1-16.

```
PATTERN  MIX    Voice=016[--:VocalTrack]
PART     1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
         DR BA KB GT AP AP AP AP AP AP AP AP AP AP AP --
VOICE NUM  1 105 57 12  1  1  1  1  1  1  1  1  1  1  1 16
BANK MSB  63 63 63 63 63 63 63 63 63 63 63 63 63 63 63 63
BANK LSB  32  1  0  8  0  0  0  0  0  0  0  0  0  0  0 60
         PD P-2 P-1 Us1 P-1 P-1 P-1 P-1 P-1 P-1 P-1 P-1 P-1 P-1 P-1 MV
VOL/PAN  VOICE [ EFFECT ] EF SEND [ VCE ED ] PLG1-3
```

If you load the SONG individually or if you load the PATTERN individually your sample will load with it. Even if you load other Voices into the internal USER 1 and USER 2 banks your songs and patterns will still load all the Voices that are used in them. Cool!

<div align="right">

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Technology Products
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