



MOTIF XS

POWER USER

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 XS. With expansion capability of 1GB, you have more than 100 minutes of stereo record time at the highest resolution (44.1kHz), that's over 200 minutes of mono recording. This means a serious amount of recording can be done. The Motif XS is a **sampler** however, and a sampler is different from a linear hard disk recorder in many ways. They both are tools for music production via recording audio. However, a sampler uses MIDI events to trigger playback. In a sampler your audio is recorded to volatile USER RAM, as opposed to a fixed hard drive. In a hard disk recorder your audio is printed to the disk. Therefore access to the data is quite a bit different. Advantages that a sampler has over a hard disk (HD) recorder are not measured in record time (the HD recorder easily wins that contest every time), but in how you can manipulate, trigger and utilize the audio data that you record.

It should be mentioned here there is a 'limit' to the longest single continuous sample that you can record - 6 minutes and 20 seconds (380 sec). This is a 64MB stereo sample at the highest resolution, 44.1kHz. This should not be a limitation really as most songs are not this long, and besides continuous recording is certainly not the most efficient way to use a sampler. Recognize that you cannot open a mic and record for 50 minutes straight...if that is what you think you need, we can recommend a tape recorder or a hard disk recorder like the AW1600!!! This is the reason we stress the point that this is sampling

and the art of sampling is quite a bit different from the art of hard disk recording.

In this article we will concentrate on the basic recording process. Because one of the innovations of the Motif-series is the way you can "overdub" audio to preexisting music - even though we are "sampling". Prior to the Motif-series, samplers would allow to work with user recorded music clips, but you were working around the audio. By this I mean you imported or recorded a sample, and then you built your composition around that sample. We can now record our sample as if we were overdubbing audio. We can record music tracks first, and then overdub a sampled vocal or a guitar solo, while listening back to previously recorded tracks.

We will concentrate on setting up a microphone as the example for this article. You can sample to SONG mode; you can sample to PATTERN mode Sections; you should figure out what will be the most convenient mode to work in.

When you are recording audio to the Motif XS you will be creating a USER SAMPLE VOICE and having the Integrated Sampling Sequencer (ISS) place a trigger event (Note-On) on the track to precisely trigger playback of that audio. Later, you can import both the MIDI data and the audio data associated with a Motif XS Song directly into Cubase - where you can add additional recording and prepare your composition for the final product. The "Extensions for Steinberg" (available for download at www.yamahasynth.com) adds the ability for any of the Cubase packages to IMPORT MOTIF XS SONG files.

What's covered?

Many times you might work on a section of music at a time. For example, working on a guitar part for the chorus section or working on the vocals of a verse. Sometimes it might be better to wait until you transfer your Patterns to a linear Song and then add the vocals or guitar as an overdub. But recognize that you are "sampling". The Integrated Sampling Sequencer of the Motif XS will allow you to sample as if you were overdubbing to a recorder; you use a punch-in/punch-out function to **optimize** your record times. It is simply wasteful to place the sampler in record and then start playing or singing 8 or 10 measures later. Therefore, learning about **punching in** and **punching out** is a very good thing. It is essential to make your samples useful. We cannot repeat enough samplers differ from hard disk recorders.

As a point of reference: Audio is recorded at a pace of 10MB per stereo minute when you sample at 44.1kHz (highest resolution). Therefore you must spend your record time wisely.

If you have never worked with any kind of multi-track recorder this may be a new concept for you. But what it means is that in order not to record "empty" air, you can be efficient by engaging the record button *just* prior to the meaningful data. In other words, if the vocals do not come in until measure 17 of the song, do not record 16 measures of empty space – it is simply a waste of precious memory. (And besides if you need to trigger this bit of audio again you don't want to have to wait for 16 measures of dead air to pass). Instead have the Integrated Sampling Sequencer "punch-in" (engage the recording) precisely at measure 17. And if the first verse is over at measure 32 and the vocals do not come back in until measure 49, you can have the ISS automatically "punch-out" at measure 33. Each time you punch-in and punch-out a Note-on event to trigger that piece of audio is automatically created and placed on the track as MIDI data. As we will learn in our next section, only MIDI data actually gets recorded to a track, in a sampler. The MIDI data triggers the audio playback. This is an important concept to understand before proceeding into the world of 'sampling'.

This article assumes that you have properly installed a pair of matching DIMMs and have verified that they are working properly. This article will cover the following topics:

- _Understanding Audio Sampling: What is it?**
- _The INTEGRATED SAMPLING SEQUENCER**
- _The Concept of 'OVERDUBBING' Audio**
- _Sampling Audio to a Motif XS Song**

- Set the A/D Input
- Set the Metronome
- Setup Audio PART
- Setup the Sampler
- Playback the Sample Voice: Confirm
- SETUP for Overdubbing additional data

_Sampling Audio to a Motif XS Pattern

- Set the A/D Input
- Set the Metronome
- Special Considerations when sampling Audio in Pattern Mode

_Setting up a Compressor

_Making your sample fit the Mix

_Creating a USER VOICE

_Creating a MIX VOICE

Understanding Audio Sampling: What is it?

"Any track can be MIDI...any track can be audio."
This is a true statement for the Motif XS; however, it is also a trick statement. When you fully understand the way the Motif XS works you will know what I mean. What the statement is trying to say is that there are 16 tracks in the Motif XS sequencer and each track can be used to contain either normal MIDI data (playing the keyboard) or can contain audio data (samples). So-called "MIDI" data tracks will have a yellow 5-pin MIDI icon, while Audio tracks will have a blue wave icon.

It is also true that Audio samples are assigned to a MIDI note (KEY BANK) 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 XS, 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 Sample 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 XS, 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 in RAM).

ROM is Read Only Memory – and refers to the type of memory where the preset waveforms are stored at the Yamaha factory. You cannot destroy them or lose them as they are permanently stored and your VOICES can access this data (read) it.

RAM is Random Access Memory – and refers to the type of memory you can add to your Motif XS (DIMMs). It is random access because you can change it. This memory is volatile (which means it disappears when you power down) but it has the advantage of being able to be rewritten. So you can add your own sample data to this memory.

Like in your computer you have ROM and RAM memory. The RAM area is where you can work and be creative. However, just like the Motif XS, the RAM area of your computer is volatile. When you power down if you did not save your data, it will be lost. As I type this article into my computer, I am working in RAM. If I forget to save and shut off my computer, oops! All my work will be lost.

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.

INTEGRATED SAMPLING SEQUENCER (ISS)

It will help to have a basic understanding of how the Motif XS 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 composition 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. Later, Pattern Sections can be linked together to create a linear Song. It is only a matter of workflow preference.

A 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. This includes any sampled data, as well. After all sampled data is contained in a User Sample Voice – so in order to move the sampled data between modes only the Voice and its trigger notes need be copied by the sequencer.

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 and based on what you need to accomplish. In our first example we will be using the Motif XS 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 a relatively new concept** for synth workstation keyboards. Before the Motif-series, "sampling" was an activity that you did without listening to anything. You, more or less, created an audio sample and then fit your music to that sample. Now we can sample data like you would with any multi-track recorder – this is why we can use the word "overdub" – typically a word used to describe adding a track in a multi-track audio recording situation. 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 a linear bit of audio (naturally). This is not a problem because as we will see a little later in the Motif XS' Pattern sequencer, each recording (called a Phrase) can be set to any length you need prior to recording (up to a maximum of 256 measures). If you want to record a 16-measure audio track over a 4-measure repeating section, you simply need to open a 16-measure phrase on your audio track. Your backing tracks will repeat 4 times while you sing/play the 16-measure phrase.

This is an important concept to understand: you can determine what length of audio you wish to record and have the ISS punch you in and punch you out. The terms are similar to audio recording devices but we are actually *sampling*. This is the innovation of the Motif-series. It allows you to work with a sampler yet use it with the same ease as you would any multi-track recorder. Behind the scenes it is working to make the sample function properly. Although we use the term "overdub" you are simply sampling audio, and the Motif XS handles the rest of the functions.

'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 4 part vocal recording. We will build these up one recording at a time – a single singer but we will make 4-part vocal 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 into a Voice. A Voice has 128 places, called "Key banks" (notes C-2 through G8) on which you can store samples (actually each note can store

several samples that can be played at different velocities – but that’s another story). What we will be doing here is setup for our first pass overdubbing to the music tracks - recording the first vocal part on note C3. Then we will playback the data on C3 and overdub a second harmony part on C#3. Then playback the first and second harmony vocal from C3 and C#3 while we record another part on D3 and so on. These will all be assigned to a single track ...just simply different notes on that single track. Four samples in the same WAVEFORM.

Each note of the sample Voice can be triggered individually. 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 XS Song

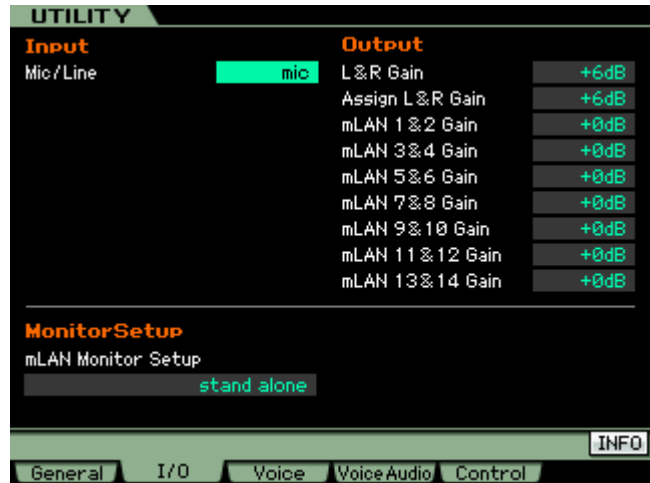
The Preparation:

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 XS (it does not have a phantom power supply). If you have a condenser microphone you will need to provide it the proper power before plugging the audio to the XS (consult your microphone’s Owner’s Manual for details). You can route the condenser microphone through a mixer that provides the +48Volts necessary, then route the output of the mixer to the A/D Input of the Motif XS (you would then set the Mic/Line setting to Line). You will want to have a good pair of comfortable **headphones** available during the overdubbing session. You will need to mute your monitor speakers or turn the amp volume completely down so that your speakers do not ‘leak’ into your recording (sample). “Leakage” is any unwanted audio getting into the recording. Therefore, you must take care to isolate your microphone from the output of your speakers. This is standard operating procedure in any recording environment. And is the reason they construct a separate room for musicians and recording engineers in any professional studio. When recording at home many times you may not have this luxury – as your control room and your studio room may be one and the same. If you are not the singer you can have one set of headphones directly from the Motif XS and the other set plugged into your mixer or, if you are not using a mixer, you can use a headphone splitter.

Set the A/D Input

- Press [SONG] - recall your composition
- Press [UTILITY]
- Press [F2] I/O

- 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. For now set it so you can hear yourself. We will make exact level setting when we get to the sampler (meter). For a microphone it is typical to have the Gain control set to maximum.



Set the Metronome (if necessary)

- Press [SEQ SETUP] – you must be in either SONG or PATTERN mode for this button to function.
- Press [F1] CLICK – set your options for the metronome.

You can select from between 10 different click TYPES. Set the “Sample Count-In”¹ measures if you need a count-in prior to sampling. (This is particularly useful 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.

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

Make sure the [TRACK] button is lit and press the appropriate Track Select button, 1-16, to select a blank track for the vocals.

Setup Audio PART with Insertion Effect

If you want to record with the INSERTION EFFECT, let’s set that up now.

- Press [MIXING]

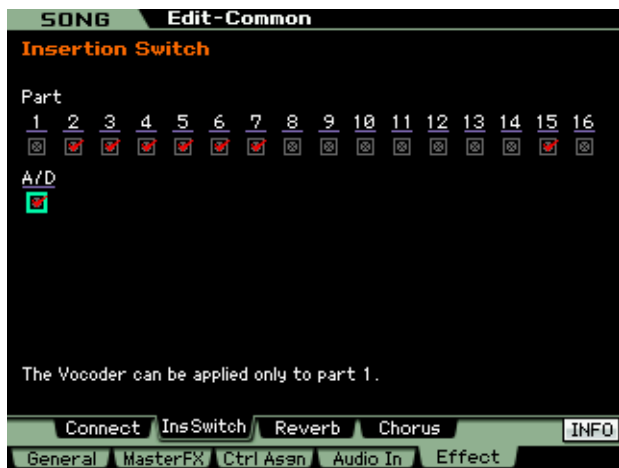
¹ The Sample Count-in will be a pre-count and will disappear once the sequencer starts. This function is very useful if you have set the record start point close to the beginning of actually music play. The pre-count can be from 1 to 8 measures in length.

- Press [EDIT]
- Press [COMMON EDIT]
- Press [F5] 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 A/D 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 Parts can have their Dual Insertion Effects active at one time. Later you can turn it back ON for the Part from which you borrow it. Once we record through this Insertion Effect the results will be recorded thus freeing the Effect for other duties.



Notice that you are placing the INSERTION SWITCH on the A/D (analog-to-digital) input PART, which will allow you to record it...

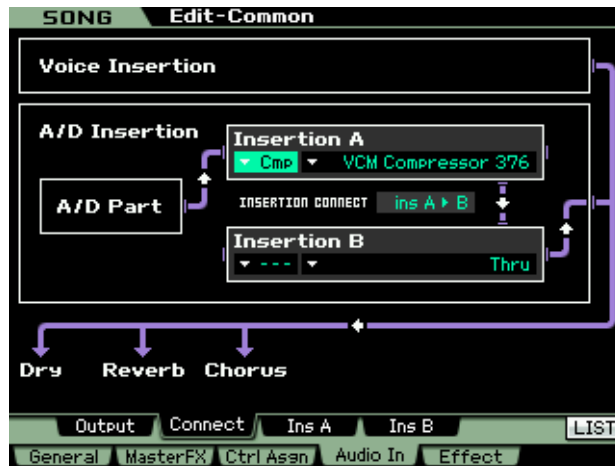
Select the INSERTION EFFECT for the A/D INPUT:

- Press [F4] AUDIO IN
- Press [SF2] CONNECT (shown below)

Here you have access to the dual processors, INSERTION A and INSERTION B. The 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.

Below, INSERTION A is set to the "Cmp", the Compressor, category and the **VCM Compressor 376** type is selected.

- Press [SF3] INS_A to edit the selected Insertion Effect parameters.²



The VCM Compressor will allow you to maximize record level and add **presence** to your recording.

- Press [SF1] OUTPUT - Set the parameters for the A/D PART, including the pan, 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 XS for the microphone input.
- Increase the Volume as necessary for listening comfort.
- The Rev_Send and Cho_Send (SYSTEM EFFECTS) **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. Or you can simply add reverb later (on playback, after you are finished sampling). Technically, we are patching the INSERTION EFFECT on the input stage of the A/D prior to the recorder (sampler). The Reverb and Chorus processors are SYSTEM EFFECTS and are located 'post' the recorder and therefore are not able to be sampled (or heard) when we enter the sampling mode.

Setup the Sampler

- Press [INTEGRATED SAMPLING] You arrive at the SAMPLE SELECT screen. It is from here that you can go in several directions. The Sample - Select screen is considered "home base" when you are in the sampling mode of the Motif XS. You can launch in several

² See the Section on "Setting up a Compressor" (page 10) for details on working with dynamics processing. Dynamics processors like the compressor are Leveling Amplifiers and can bring your vocal recording to life.

directions from this screen. The first parameter on the SELECT screen is the target WAVEFORM number 0001 – 1024.

We want to go to RECORD:



- Press [F6] REC> – You will arrive at the Record Setup screen (below). Here we will tell the unit where the sample is coming (Source) from and what type of sampling we are going to be doing. We will also set where it is going (Destination).
- Set the RECORDING TYPE to 'sample+note'
- INPUT SOURCE is **A/D INPUT**
- Set the Mono/Stereo to 'monoL+R'

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

- Set the Destination parameters on the right side of this screen



Since we have selected "Sample+Note" as the sampling TYPE you can assign the target Track (16) and can target a specific Key bank (note) for

the MIDI trigger event (C3). C3 is the default Key bank assignment and refers to the note middle "C". ("Part", "Voice" and "Drum Key" will only be available when you are sampling in other modes).

Sample+Note RECORD TYPE, means literally that the sample will be recorded and that a MIDI Note-On event will be created on the sequencer track to trigger playback of that audio. The duration (or "gate time") and placement of the note will be determined by the TRIGGER Mode selected in our next setup step. The creation of this note-on data is critical to the playback of the audio you record – by precisely placing this note-on trigger event the ISS ensures accurate playback timing of your audio with the sequenced tracks to which you are 'overdubbing'.

- Press [F6] STANDBY – To setup for the actual sample method (Trigger mode). **Trigger mode** is simply "HOW" the sampling will begin.

While in STANDBY you can test the microphone and see the graphic bar meter that represents the record level. The object is to get as much level as possible without clipping. The more level the better – short of distortion.³ Adjust the GAIN control on the back panel for more or less level. The "Record Monitor" parameter is set for your listening comfort and does not affect the recording level (at all).

- Select a TRIGGER MODE. Here are the three options:

Level – sampling begins when audio exceeds a particular setting value. Sampling ends manually by pressing [F6] STOP

Measure – sample begins automatically when the sequencer play reaches the Punch-in point and ends automatically when it reaches the Punch-out point

Manual – sampling begins when you press [F5] START and ends when you press [F6] STOP

How to Choose a Trigger Mode (wisely)

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.

³ Distortion should be determined by both your eyes and ears. If the meter says the signal is fine and your ears tell you it is distorted, **always** believe your ears.

Level would be useful if you are not sure when the audio is going to begin. As soon as the signal passes the pre-determined level, sampling begins. There is always 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 [F5] START to arm the sampler, and then press play on the CD player, the Motif XS screen will read, "Waiting" until the level reaches the trigger point – then 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.

Measure is used when you know the punch-in and punch-out points by measure. This is useful for most sequencer (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 [F5] START, the screen will read "Waiting". It is 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. This is similar to how a multi-track tape or a hard disk recorder would allow you to setup in advance where the recording is to begin and end. And as you can see for recording vocals or guitars, etc., this is ideal. It will also make triggering and editing of the sample very easy as you know the punch-in and out points are aligned to your measure lines.

Manual is used when you want to control the start and stop of the sample yourself. Sampling begins when you press [F5] START 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 MEAS (MEASURE) Trigger Mode. Determine at what measure you want to punch-in and punch-out.

In the next screen shot we are setup with the Trigger Mode set to Measure. We are set to punch in at the top of measure '009' and punch out at the top of measure '017'.



You see we are targeting, Waveform 0001, track 16, the transport of the sequence is currently at measure 005, and our Key (Key bank) is C3. We are set to record an 8 measure phrase 009-017 into a 16-measure Section. It is important to realize that to record measure 016, you must set the punch-out point to the top of measure 017.

- Press [F5] START...
- "Waiting..." appears. It is waiting for you to start playback.

The current measure reads out in the upper right corner. Use the [>>] FF or [<<] RW buttons to locate a comfortable start point prior to the punch-in point. We are set to Measure 005 meaning four measures will play before the punch-in occurs.



Press [>] the Start button on the sequencer transport... Playback and the sampling will begin, automatically, when the sequencer passes the punch-in point at measure '009' and will automatically punch-out at the beginning of measure '017' and then STOP.

If you need to start the sample at the very top you can use the special Sampling Count-in feature.

Playback the Sample Voice:

When sampling is complete you will be given the opportunity to AUDITION the recording via the [SF1] button. The sample is not kept unless you approve of it and press [ENTER].

- “Press [ENTER] to fix”
Therefore auditioning it is very important.

Important note: Expect that the volume of the sample will be lower than what you hear going into the sampler – this is normal.⁴ If you are confident that your GAIN level meter setting was good, you can deal with balancing the playback volume later. The playback volume will be under the influence of several other factors which we will program later. Your initial job is to ensure good record meter readings and to worry about the actual musical performance. We can deal with playback volume and mixing, later.

- Press and hold the [SF1] AUDITION button to hear your recording playback (isolated). You should not hear anything but your recorded sample (if you hear the rest of the music backing tracks, this means you forgot to turn off your speakers). You want to isolate (as much as possible) the incoming signal.
- Press [ENTER] to accept the results or [EXIT] to discard the audio and try again.

When you press [ENTER] you have created a special User **Sample Voice** that is ‘local’ to the current Song or Pattern. It will automatically be placed in your MIXING setup. It will reside in a special bank “63/50” for USER SAMPLE VOICES that reside within the sequencer. And as this was the first sample in this Song/Pattern it will be designated as User **Sample Voice 1**. The Integrated Sampling Sequencer will have created a note-on on the track with a precise duration (gate time) equal to the number of beats from the punch in point to the punch out point. And a blue waveform icon will identify this track (as opposed to the green-yellow MIDI jack icon found on MIDI tracks).

⁴ Playback volume will be significantly lower than the Voices of the Motif XS. All Motif XS Voices have been programmed – meaning they have been setup for velocity response, EQ, etc. Your recording is a “naked” audio clip in comparison. Simply optimize your record level and deal with playback volume later (that is a matter of mixing). You are listening to the **input** source prior to sampling and you are listening to playback of the sample afterwards.

The default Velocity is 100. The default Voice Volume is 100. If your sample is not loud enough (which is quite possible as it is just a raw sample) you can increase the Velocity of the note on and the Volume of the track’s PART:

- Press [SONG]
 - Press [EDIT] to view the Event List
 - Use the cursor arrows to navigate this screen
 - Set the Velocity to 127
 - Press [ENTER]
- Press [MIXING]
 - Raise the CS for Track 16 so that the volume is set to 127, or as necessary
 - Press [STORE]/ [ENTER]/ [YES] to store the Sequence and Mix data



- Note: In the screen above a note-on event was created on the 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 100. The Velocity (VEL) determines how loud the audio clip plays back. “TOP” denotes the beginning of this track and “END” denotes the end of that track – so this note-on event and the PC or Program Change is currently the only events on that track. The PC event is automatically inserted to ensure that the Sample Voice that was created is recalled to playback this data. (Bank numbers 063-050 and the first PC is 000)

This is “Sample+Note” in action. If you used the VCM Compressor 376 your sample will probably not need to be boosted. However, your mileage will vary.

Of course, mixing vocals (or any data that you have sampled) in with your pre-recorded tracks, is a matter of balance. Do not be afraid to turn

things down to accommodate your vocals. The difference between a musician and a recording engineer – is the recording engineer knows how to turn something down! Musicians, in general, only know how to turn something up. I’m not saying this to put musicians down (I’m both a musician and a recording engineer). Those who know me, know that I was a musician who learned all about recording engineering. (Actually I eventually became a recording engineer) My first experience in the studio as a musician made me realize how important it was to learn that set of skills that would allow me to communicate with the recording engineer.

Special Considerations when Sampling Audio to a Motif XS Pattern

Nothing prevents you from sampling into a Pattern Section. However, the fundamental difference between Song mode (linear) and Pattern (cycling Sections) will cause a few differences in how you approach the task. There is no one way to work. However, Pattern mode is designed as a construction tool and you may want to wait until you have completed the construction of your composition before recording (sampling) audio. This is why the article covered SONG mode sampling first. But nothing says you have to sample exclusively in SONG mode. In fact, if you understand what is going on and are aware of the rules of PATTERN mode, nothing prevents you from sampling into a PATTERN SECTION.

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 set 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 then convert the Chain to a SONG. This could be due to things like musical pickups (where the audio begins prior to the downbeat) or overhangs (where a note is held into the next section)⁵. It will be a decision that you have to make based on the composition you are working on at the time. If you do determine that you want to sample into a Pattern please understand that sampling is still 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:

⁵ This would be a good reason not to sample in PATTERN mode. Patterns have a definitive start and end point.

- Press PATTERN to view the main Pattern screen
- Move the cursor to the right and highlight the **Length** parameter (upper right corner)

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 LENGTH 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 a 16-measure vocal phrase. (Any length can be used up to a maximum 256 measures). Because you are sampling and not recording audio as on a tape recorder, you want to do some planning. It would be very wasteful to record 8 measures of empty data if the vocal begins at measure 009, and it would be wasteful to let empty bars record if the vocal did not occur between measures 33 and 65. You are sampling so spend your memory wisely. Plan ahead. A trigger note must be used to trigger playback. If you know this you can plan ahead and set your record points at convenient musical locations... like the beginning of a verse, the beginning of the chorus, etc.

You can use the punch-in and punch out feature to optimize the use of your sample memory.

Important note: In order for all phrases to play properly, quite naturally the LENGTH parameter should be set to match the length of the longest phrase in the Section. If the LENGTH parameter is set shorter than the longest phrase, all tracks will be shortened to the set length.

Setting up for sampling into a PATTERN is exactly the same as before.

Set the A/D Input

- Press [PATTERN] - recall your composition



- Press [UTILITY]
- Press [F2] I/O
- 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. For now set it so you can hear yourself. We will make exact level setting when we get to the sampler (meter). For a microphone it is typical to have the Gain control set to maximum.

Metronome Considerations

- Press [SEQ SETUP] (You must be in either SONG or PATTERN mode for this button to function).
- Press [F1] CLICK – set your options for the metronome.

Set the **Sampling Count-In** if you need a count-in prior to sampling. (This is particularly useful in Pattern mode if your sample must begin at the downbeat of the first measure.) The click will count-in and then stop (obviously you don't want to record it along with your sample). 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.

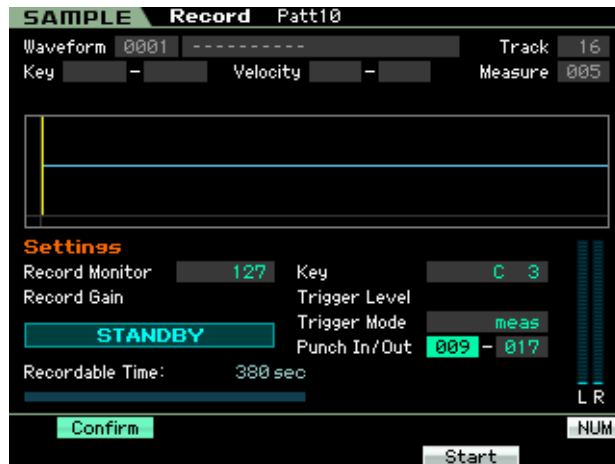
Make sure the [TRACK] button is lit and press the Track Select button, 1-16, to select a blank track for the vocals.



Setup for sampling as before: Using "sample+note", "A/D Input" as the Input Source; "monoL+R" as the Stereo/Mono setting, and target the Keybank and Track, as necessary. If you need the details please see the previous section's SONG mode example. This portion of the setup is exactly the same.

- Press [F6] Standby

This will move the screen where you can select the TRIGGER MODE.



Select **Measure** as the TRIGGER MODE and set the Punch-in and Punch-out points. The Punch-out point can only be set to maximum value determined by the Section Length. The "Length" parameter determines the loop point for the Pattern Section is determined prior to recording in PATTERN mode. Remember, the difference between Song mode and Pattern mode, is that Pattern mode has a set start and end point.

- Press [F6] REC to see "Waiting.."
- Press [>] To begin sequencer playback

When the sequencer reaches the punch-out measure sampling will stop automatically.

Planning Ahead if using PATTERN mode:

Since recording (sampling) into a PATTERN Section presents some different challenges from sampling while in SONG mode, we recommend that you plan accordingly. In a SONG you have just one TRACK 1, however, in a PATTERN you have 16 TRACK 1's. What? That's right, each of the 16 Sections of a PATTERN has its own Track 1. This will need to be considered when sampling into PATTERN mode. If, for example, you are going to be recording your vocals to several different Sections, you may want to include all of your vocals in the same WAVEFORM – this is logical. For example, say Section [A] is the first verse and Section [B] is the second verse. It will be wise to target KEY BANK "C3" in Section [A] and then target the next higher note, KEY BANK "C#3" for Section [B] – keeping both samples in the same WAVEFORM.

When you are ready to record the Chorus section to Section [C], you can move to the next higher

note, "D3", and so on. Although you could use a different WAVEFORM for each recording (one per Section) this would not only be wasteful from a memory standpoint (there are *only* 1024 WAVEFORMS) but if you use a different WAVEFORM on the same TRACK number – you will have to deal with Program Change data being inserted at the top of each Section. You should avoid this whenever possible. This will be unnecessary if you simply plan ahead and target the next higher Key bank for each successive section. This will also make life much easier if you later want to apply Effects, EQ and otherwise edit the Sample Voice (see the end of this article for tips and tricks on converting your audio into a full-fledged Motif XS VOICE and into a MIX VOICE).

If you are attempting to create multiple harmony parts, as we did in the SONG portion of the tutorial, you will need to target different areas of the keyboard for each Section. For example, use different octave ranges per Section. Use the octave between C3-B3 for Section [A], and C4-B4 for Section [B], and C5-B5 for Section [C] and so on. This way all your vocals can be in the same WAVEFORM... remember there are a total of 128 places to store audio in any User Sample Voice. Keeping all your samples that are of a similar type in one waveform is not only economical it is practical from a standpoint of mixing. So if you are working with several audio overdubs, keep the vocals in their own waveform, keep the guitar overdubs in their own waveform, etc.

It is like each audio track has 128 virtual audio tracks that you can use simultaneously (not that you would, but you could).

At any rate you do not want to target Key bank "C3" for all of your recordings because a key bank can hold a maximum of two samples at the same velocity. If you target different Key banks you realize that you can keep several "takes" and later choose the best one. For example, singers often try a pass, and want you to keep the take, while they try another. When you understand what the ISS is doing, you realize that you can edit the Note-On data (set the velocity of the note-on) so that you control what key bank is audible. Target the next higher note and let the singer try another take. By adjusting the velocity of the Note-On you can determine which key bank is going to playback. You can go to the [PATTERN] or [SONG] PLAY screen and press [EDIT] to view the Event List. Setting the Velocity of a particular note-on to a velocity of 001, will make it virtually in audible... in this manner you can effectively temporarily mute a previous take. If later you decide it was the "keeper", you can simply restore its velocity.

Advanced: 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 audio up to theoretical maximum – it also brings up any background noise (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. We recommend that you get the record signal right the first time so that "normalizing" is unnecessary! If you optimize your record level in the first place, normalizing is totally unnecessary. A compressor is a "leveling amplifier" and can be put to good use in the sampling process – particularly when sampling vocals.



When recording vocals in any professional recording studio, a processor called a *compressor* is often used. A compressor is in a category of tools called Dynamics Processors. 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 **input** signal exceeds the threshold (level) the compressor will automatically reduce the level. This is called 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 distance (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 'having presence' ...this calls for the artful use of a compressor. 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

beyond serving as a starting point). 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:

Attack - Attack controls the first few milliseconds of the signal (0.022ms-50.40ms). Attack time will allow this amount of 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 (10.88ms-544.22ms). 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. It is more a thing where you will know when you have it wrong - because one of the goals of good compression is that you are typically unaware of it, unless something is overdone.

INPUT: Threshold/Gain Reduction - 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. 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. In many classic compressors the

INPUT parameter controls both the Threshold and influences the amount of Gain Reduction applied to the signal. Please see the Power User article on the **VCM COMPRESSOR 376**. A very percussive sound like a snare drum hit can have a transient peak (quick level spike) that can average about 14dB above the body of the sound. By reducing this spike via the compressor you can have much more impact for the response of the drum. As you gain more experience working with incoming signals you start to be able to analyze what signals have these big transient spikes that can prevent you from getting a good 'presence' when you attempt to record them. Within the VCM COMPRESSOR 376, the INPUT signal will affect not only the amount of incoming signal but it will determine the amount of Gain Reduction. As you raise the input knob you will see more negative deflection (in response to signal) of the VU meter. The VCM Compressor is a model of a "fixed threshold" type compressor. This type makes it much easier on the engineer (you) in finding and setting the point at which the reduction of gain begins.

Ratio - Ratio is the degree that the compressor is working. The ratios are always to 1. Ratios of 2.0 (two-to-one), 4.0 (four-to-one), and 8.0 (eight-to-one) are appropriate for vocals. 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 wanted 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. Don't let the math hurt your head... the thing is the compressor holds the signal back from going too far. The difference between a Compressor and a Limiter is one of degree (10:1 and above are considered 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 (∞ - 0.00dB). 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 parameter 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's 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 parameter would allow you raise the overall signal back up. So the whisper will sit on top of and be heard in context of the mix and 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. Remember 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 actually hear the compression, you have too much". No discussion of compression would be complete without a warning: do not compress everything. Try and use it with good sense. What is more important than LOUD in any good mix is the sense of soft-to-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 the 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. So look for negative deflections on the extremely loud peaks - not constantly.

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 XS file architecture and can be loaded individually (File Type: SAMPLE VOICE) from within any ALL SONG (.x0s) if you recorded to a SONG, any ALL PATTERN (.x0p), if you recorded to a PATTERN, or any ALL data (.x0a) file.

Creating a Normal USER VOICE

As an alternative method you can create a normal USER Voice from your recording. This will place it in the general Voice pool; bank 63/8, 63/9 or 63/10 (User 1, User 2 or User 3) and will allow you to fully edit the sound like any other Voice in the Motif XS. Initialize a USER VOICE and assign your Wave to it. Here's how:

- Press [JOB]
- Press [F1] INIT
- Press [ENTER]
- Press [INC/YES]
- Press EDIT
- Press Track Select [1] to select Element 1
- Press [F1] OSCILLATOR point the Voice's WAVE BANK 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 create 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. This means you can add envelopes, filters, the Dual Insertion Effects, EQ, etc., etc.

Located at [F6] EQ 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. If you are not getting enough level compared to your tracks, a +6 or +12 dB boost will place your sample out front. It is rarely, if ever, necessary to set a +18dB boost (if you find this necessary you should probably re-record your data as this is a tremendous amount of level).

- Name and Store it to a USER location – it will be in bank 63/8 (USER 1), 63/9 (USER 2) or 63/10 (USER 3).

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. Then [STORE] your mix.

- Press [MIXING]
- Press [EDIT]
- Press the Track that contains your new USER VOICE
- Press [F5] VCE EDIT

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; as long as the PART contains a non-drum kit voice.

- [STORE] the Voice to the MIX VOICE location

The advantage of placing your User Voice in the 63/60 MIX VOICE bank is that – being “local” means it will (again) always load when you load the SONG or PATTERN that it is in. Meaning if you want to load the Song or Pattern as in individual entity, your sampled audio will load as well. The MIX VOICE bank can hold a single Normal (non-Drum Kit) Voice per each Part 1-16.

If you load the SONG or PATTERN individually your sample will load with it. Even if you load other Voices into the internal USER 1, USER 2 and USER 3 banks your songs and patterns will still load all the Voices that are used in them. Cool!

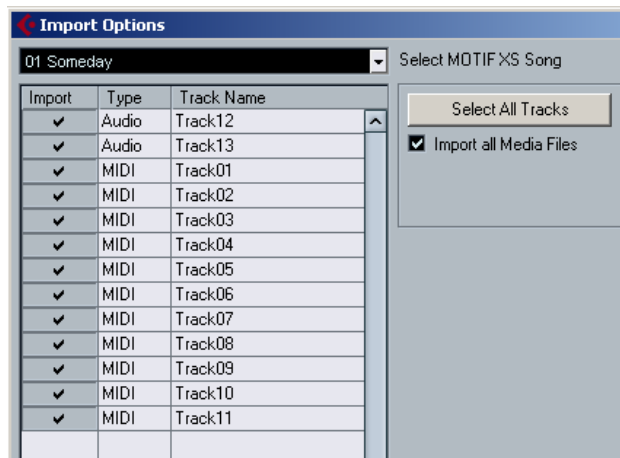
IMPORT MOTIF XS SONG – Cubase/Nuendo

Added to Cubase Studio 4, Cubase 4, Nuendo 4 and the introductory version Cubase AI4, is the ability to IMPORT Motif XS SONG data directly into a Project on your computer. This is the type of tight integration between hardware and software that will set the standard for future applications. Here’s how it works (in theory):

You begin your project in the Motif XS sequencer. The Integrated Sampling Sequencer of the Motif XS has many powerful tools for music production. You may, for example, start in PERFORMANCE mode with a germ idea (a music motif, which you build into a full composition). Using the Direct Performance Record function you transfer the basic idea into the sequencer. Direct Performance Record allows you to perform up to four Parts of music simultaneously via live playing and with the powerful four-Part arpeggiator. It can be used to transfer ideas to a linear SONG or to a PATTERN Section. SONG mode can be used to freely play/record and later “cut and paste” your best

ideas into an arrangement. PATTERNS can be used to work musical Sections into an arrangement – trying out different configurations and play orders. Finally you can Chain Pattern Sections into a complete linear arrangement, add a vocal track or overdub a live guitar.

Once you have a complete linear Motif XS SONG, you would [STORE] that to Flash ROM, then [SAVE] an ALL data file (.X0A). An ALL data file quite literally saves all the data currently in your Motif XS – this includes all your Songs, Patterns, User Voices, samples, arpeggios, Performances, Master setups, etc., etc., etc. It is ALL data. This file can be accessed by Steinberg DAW software packages using the FILE > IMPORT function. Once you have selected a Directory for your new Project, the DAW will show a dialog box where you can select from the SONG titles contained in your ALL data file (shown below)



Select the tracks you wish to import or click the “Select All Tracks” – a check mark will appear next to the Track in the IMPORT column. You can check the “Import all Media Files’ option to ensure that all samples associated with the Song are imported to the Audio Pool of your DAW.

Once the data has been imported, simply launch the STUDIO MANAGER: **Motif XS Editor** and select the BULK option on the main menu bar. Select the RECEIVE Current Mix option (and User Voices, if necessary) to capture the all your XS settings. Now when you save your Steinberg Project not only your sequence data (MIDI and audio) but your entire Motif XS mix will be completely captured. Very Cool!

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